NEC

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NEAX®2000 IPS

(INTERNET PROTOCOL SERVER)

General Description

February, 2005

NEC Unified Solutions, Inc.

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NEC Unified Solutions, Inc.

NEAX2000 Product Management

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Regulatory Information

Regulatory Requirements

The Federal Communications Commission (FCC) has established rules that permit the PBX to be directly connected to the telephone network. A jack is provided on party lines or coin lines.

The telephone company may make changes in its technical operations and procedures. If such changes affect the compatibility or use of the PBX, the telephone company must provide adequate notice of the changes.

This equipment complies with the requirements in Part 15 of FCC Rules for a Class A computing device. Operation of this equipment in a residential area may cause unacceptable interference to radio and TV reception requiring the operator to take whatever steps are necessary to correct this interference.

FCC Part 15 Requirements

In compliance with FCC Part 15 Rules, the following statement is provided:

Warning: *This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference.*

FCC Part 68 Registration

Company Notification

Before installing the PBX to the telephone network, the telephone company must be provided with the following:

- Your telephone number
- The FCC registration numbers:

	JAPAN	USA
PBX	AY5JPN-20542-PF-E	AY5USA-21582-PF-E
Hybrid	AY5JPN-20543-MF-E	AY5USA-21583-MF-E
Key System	AY5JPN-20586-KF-E	AY5USA-21584-KF-E

The Ringer Equivalence Number is 1.6B; the required USOC jacks are RJ21X, RJ2EX, RJ2GX, and RJ49C.

Note: *Limitations on features exist if the system is registered as a KF system. Refer to Features and Specifications for details.*

Location of FCC Compliance Labels

Labels stating the NEAX2000 IPS FCC registration number and compliance with FCC Parts 15 and 68 are attached on the inside of the system's front cover. Label examples are as follows:

"This equipment complies with the requirements in Part 15 of FCC Rules for a Class A computing device. Operation of this equipment in a residential area may cause unacceptable interference to radio and TV reception requiring the operator to take whatever steps are necessary to correct the interference."

NEAX2000 IPS
Complies With Part 68 FCC Rules
FCC Registration Numbers
Ringer Equivalence: 1.6B
NEC Unified Solutions, Inc.   MADE IN USA

FCC Requirements for Private Line Operations

In order to connect this system to the private line network, provide the telephone company with:

- The quantities and USOC numbers of the required jacks (See the following table.)
- The sequence in which the trunks are to be connected
- The facility interface codes by position
- The Ringer Equivalence Number or service order code, as applicable, by position

Mfg's Port ID	Facility Interface Code	Network Jacks	Service Order Code
PN-4COTB	02LS2	RJ21X	
PN-4COTB	02GS2	RJ21X	
PN-4COTG	02LS2	RJ21X	
PN-4COTG	02ES2	RJ21X	
PN-AUCA	02RV2-T	RJ21X	
PN-4DITB	02RV2-T	RJ21X	
PZ-8PFTA	02LS2	RJ21X	
PN-8COTQ	02LS2	RJ21X	
PN-8COTS	02LS2	RJ21X	
PN-8COTS	02GS2	RJ21X	
PN-AUCA	0L13A, 0L13B, 0L13C	RJ21X	9.0F
PN-20DTA	TL11M	RJ2EX	9.0F
PN-20DTA	TL31M	RJ2GX	9.0F
PN-24DTA	04DU9-BN	N/A	6.0P
PN-24DTA	04DU9-DN	N/A	6.0P
PN-24DTA	04DU9-1KN	N/A	6.0P
PN-24DTA	04DU9-ISN	N/A	6.0P
PN-24DTA	04DU9-IZN	N/A	6.0P
PN-BRTA	02IS5	RJ49C	6.0Y
PN-24PRT-A	05DU9-BN	N/A	6.0P
PN-24PRT-A	04DU9-BN	N/A	6.0P
PN-24PRT-A	04DU9-1KN	N/A	6.0P
PN-24PRT-A	04DU9-1SN	N/A	6.0P
PN-24PRT-A	04DU9-1ZN	N/A	6.0P
PN-24CCT-A	04DU9-BN	N/A	6.0P
PN-24CCT-A	04DU9-DN	N/A	6.0P
PN-24CCT-A	04DU9-1KN	N/A	6.0P
PN-24CCT-A	04DU9-1SN	N/A	6.0P
PN-24CCT-A	04DU9-1ZN	N/A	6.0P
PN-24DTA-C	04DU9-BN	N/A	6.0P
PN-24DTA-C	04DU9-DN	N/A	6.0P
PN-24DTA-C	04DU9-1KN	N/A	6.0P
PN-24DTA-C	04DU9-1SN	N/A	6.0P
PN-24DTA-C	04DU9-1ZN	N/A	6.0P
PN-2BRTC	02IS5	N/A	6.0Y

Service Requirements

In the event of equipment malfunction, all repairs will be performed by NEC or an authorized distributor. It is the responsibility of users requiring service to report the need for service to NEC or to one of their authorized distributors.

If trouble is experienced with this equipment, please contact NEC America, Inc., at 800-TEAM NEC (800-832-6632) for repair and/or warranty information. If the trouble is causing harm to the telephone network, the telephone company may request that you remove the equipment from the network until the problem is resolved.

If the equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations, or procedures that affect the operation of the equipment. If this happens, the telephone company will provide advance notice so that you can make necessary modifications in order to maintain uninterrupted service.
NO REPAIRS CAN BE DONE BY THE CUSTOMER.

Direct-Inward Dialing (DID) Calls

Allowing this equipment to be operated in such a manner as to not provide for proper answer supervision is a violation of Part 68 of the FCC's rules.

PROPER ANSWER SUPERVISION IS WHEN:

a.) This equipment returns answer supervision to the PSTN when DID calls are:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user
- Routed to a dial prompt

b.) This equipment returns answer supervision on all DID calls forwarded to the PSTN.

- Permissible exceptions are:
- A call is unanswered
- A busy tone is received
- A reorder tone is received

EQUAL ACCESS REQUIREMENTS

This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

Caution: *The use of a monitoring, recording or listening devices to eavesdrop, monitor or record telephone conversations or other sound activities, whether or not contemporaneous with its transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to the telephone conversation, such as using a beep tone or other notification methods or require the consent of all parties to the telephone conversation, prior to monitoring or recording a telephone conversation. Some of these laws incorporate strict penalties.*

Regulatory Information On Analog Telephones

NEC single-line telephones comply with Part 68 of FCC Rules. On the bottom of the equipment is a label that states, among other information, the FCC registration number and ringer equivalence number (REN) for the equipment. If requested, this information should be provided to the telephone company.

The equipment uses the following USOC jacks: RJ11C.

The equipment should be used only behind a PBX or KTS. The REN is used to determine the number of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all, areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to the line as determined by the total RENs, contact the telephone company to determine the maximum REN for the calling area.

Hearing Aid Compatibility

The Dterm terminals provided for the NEAX2000 IPS are hearing aid compatible. FCC rules prohibit the use of non-hearing aid compatible telephones.

NEC-type single-line telephone sets used in conjunction with the NEAX2000 IPS are hearing aid compatible. If other than NEC-type single-line telephone sets are to be used with this system, ensure that these are hearing aid compatible.

Industry Canada CS-03

Certification number: 140 5976 A

Load Number of the equipment: 1.0

NOTICE: The Industry Canada label identifies certified equipment. The certification means that the equipment meets certain telecommunications network protective operational and safety requirements. The department does not guarantee the equipment will operate to the user's satisfaction.

Before installing the equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions might not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or installations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request that the user disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This protection may be particularly important in rural areas.

Caution: *Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.*

NOTICE: The Load Number assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop which is used by the device, to prevent overloading. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the load numbers of all the devices does not exceed 100.

Safety Certifications

This equipment has been listed by Underwriters Laboratories and found to comply with all the applicable requirements of the standard for telephone equipment U.L. 1459. This equipment complies with Canadian Standards Association's standard C 22.2 No. 225.

Safety Considerations

When using telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock, and injury. Precautions include the following:

- Never install telephone wiring during a lightning storm.
- Never install a telephone jack in a wet location, unless the jack is specifically designed for wet locations.
- Never touch an uninsulated telephone wire or terminal, unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

Note: *More detailed precautions are included in this manual.*

Safety Instructions

1. Never install telephone wiring during a lightning storm.
2. Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
3. Never touch un-insulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
4. Use caution when installing or modifying telephone lines.
5. Read and understand all instructions.
6. Follow all warnings and instructions marked on the product.
7. Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a damp cloth for cleaning.
8. Do not use this product near water, for example, under water pipes near a bathtub, sink, or laundry tub, in a wet basement, or near a swimming pool.
9. Do not place this product on an unstable cart, stand, or table. The product may fall, causing serious damage to the product.
10. Slots and openings in the cabinet and the back or bottom are provided for ventilation, to protect it from overheating, these openings must not be blocked or covered. The openings should never be blocked by placing the product on a bed, sofa, rug, or other similar surface. This product should never be placed near or over a radiator or heat register. This product should not be placed in a built-in installation unless proper ventilation is provided.
11. This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power source available, consult with your local power company.
12. This product is normally connected with a three-wire grounding type plug, a plug having a third (grounding) pin. This plug will only fit into a grounding type power outlet. This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the grounding type plug.
13. Do not allow anything to rest on the power cord. Do not locate this product where the cord will be abused by persons walking on it.
14. Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
15. Never push objects of any kind into this product through cabinet slots as they may touch dangerous voltage points or short out parts that could result in a risk of fire or electric shock. Never spill liquid of any kind on the product.
16. To reduce the risk of electric shock, do not disassemble this product, but take it to a qualified serviceman when some service or repair work is required. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock when the appliance is subsequently used.

17. Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:
    a.) When the power supply cord or plug is damaged or frayed.
    b.) If liquid has been spilled into the product.
    c.) If the product has been exposed to rain or water.
    d.) If the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions, because improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to normal operation.
    e.) If the product has been dropped or the cabinet has been damaged.
    f.) If the product exhibits a distinct change in performance.
18. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
19. Do not use the telephone to report a gas leak in the vicinity of the leak.
20. Warning for US and Canada only

WARNING

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

This Class A digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe A est confor̀me a la norme NMB-003 du Canada.

Chapter 1 Introduction

This manual provides an overview of the NEAX®2000 IPS (Internet Protocol Server) stored program control digital electronic PBX. An introduction to the technical characteristics is included, along with a description of available system applications.

System Information - NEAX®2000 IPS

The NEAX®2000 IPS (Internet Protocol Server) is a full-featured IP based communications system providing a rich feature set with pure Voice over IP (VoIP) communications (peer to peer connections), across corporate Local and Wide Area Networks (LAN and WAN). The NEAX 2000 IPS DtermIP telephones are designed to provide a converged infrastructure at the desktop, with a 100 Base T Ethernet connection to the LAN and built-in hub for a PC connection to the telephone itself. The system can provide peer-to-peer connections between DtermIP telephones with voice compression, offering existing Dterm Series i telephone features. On the WAN side, the system can provide peer-to-peer connections over IP networks with the voice compression, on a CCIS basis (CCIS over IP) or Remote PIM (Remote PIM over IP).

NEAX 2000 Internet Protocol Server (IPS)

The NEAX 2000 IPS can provide legacy station/trunk interfaces to support the existing Time Division Multiplexing (TDM) based infrastructure, such as analog telephones, analog networks, and digital networks (T1/E1, ISDN etc.). At maximum configuration, the system can provide 1020 ports for IP and legacy devices, and 256 ports for Application cards. Communications between legacy stations/trunks and DtermIP telephones/IP networks are made via IP PAD, which converts packet-based voice data to TDM-based voice data, and vice versa. Both peer-to-peer connections and TDM-based connections are controlled the Main Processor (MP) card. The MP card incorporates a built-in Device Registration Server (DRS) and a single interface point of IP connection to IP telephone, MATWorX, and OAI/ACD servers.

NEAX 2000 IPS users have access to hundreds of service features that are used in building unique telephony applications that enhance productivity, reduce operating costs and improve communications efficiently. The innovative modular hardware and software design allows efficient, effective growth within each module from its minimum to its maximum configuration. The NEAX 2000 IPS software design is as advanced as its hardware. It ensures the system will support evolving applications and have the reliability needed to compete in today's world and into tomorrow's. The software is designed with modularity in mind. Together, these modular building blocks allow customers to initially buy what they need and add capacity and capabilities as the business demands, resulting in a greater degree of cost control for new installations and for upgrades to features, capacities and the software versions.

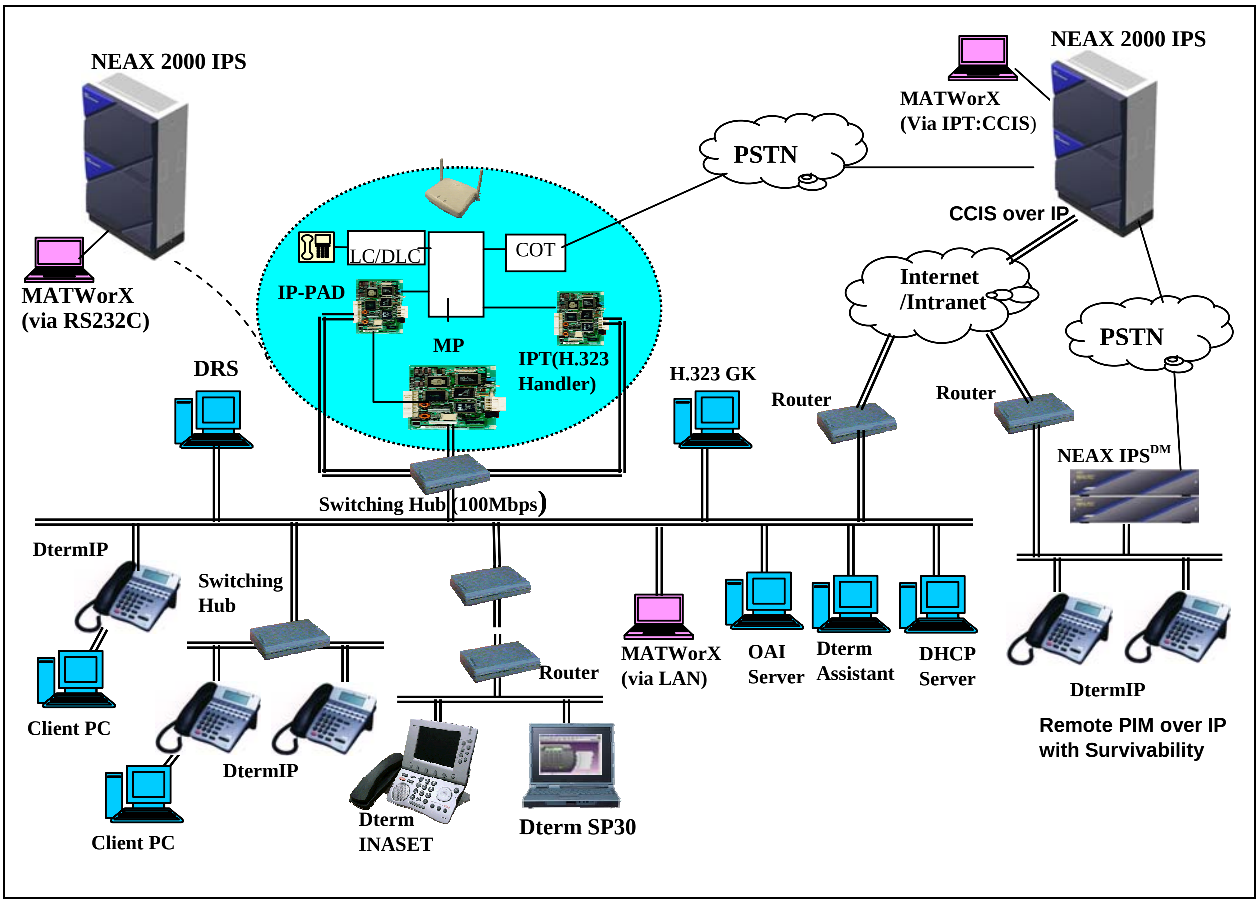


Figure 1-1 System Outline of an IPS

Hardware Architecture

Hybrid System of IP (peer-to-peer connection) and TDM Switching

The NEAX 2000 IPS supports both pure IP switching (peer-to-peer connections) and Time Division Switching (TDM). The pure IP switching is provided for communications between DtermIPs and for CCIS/Remote PIM connections with another NEAX 2000 IPS/ NEAX IPS^{DM}/2400 IPX (CCIS over IP or Remote PIM over IP). On the other hand, the TDM switching is provided for communications between legacy stations/trunks. Connections between DtermIP/CCIS or Remote PIM over IP and legacy stations/trunks are made via IP PADs, which converts packet-based voice data to TDM-based voice data, and vice versa.

Powerful, One-board Main Processor (MP) with Integrated Functionality

The NEAX 2000 IPS Main Processor (MP) is the heart of pure IP connections and TDM-based connections. The MP employs a high-speed CPU, which is equivalent with Pentium. With this processing power and System On Chip (SOC) technology, the MP integrates Device Registration Server (DRS), AP01 (OAI) functions, which are provided by an additional card in the previous IVS series. Also, by means of today's advanced LSI technology, the MP card size is minimized and On-board Ethernet Interface card is mounted on the MP without using an additional slot space in the PIM. This interface card is linked with LAN for call control processing of DtermIP and inter-work with MATWorX and OAI server.

The MP provides LAN control function, System-based Device Registration Server (DRS), Built-in FP, Built-in OAI, Built-in SMDR, Built-in CCH-IPT, 33 MHz PCI BUS, Memory (Basic/Expansion), TDSW (1024 CH × 1024 CH), 16-line CFT, PB Sender, Clock, PLO two ports (Receiver Mode/Source Mode), two RS-232C Ports, two-line DAT (Recording duration: a maximum of 128 seconds), DK, 4-line PB Receiver, Modem for remote maintenance (33.6 kbps), internal Music-on-Hold Tone, BUS Interface. BUS Interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. One card is required per system.

Reduced Hardware with IP based Architecture

The DtermIPs connected to the LAN do not require DLC cards because they can be interfaced directly with the LAN and connected with peer-to-peer basis. When the DtermIP is connected to a station/trunk that is using TSW, the speech path between LAN and TSW is made via IP PAD under the call processing control of the MP. The DtermIP can be expanded simply adding the terminal itself and IP PAD if traffic volume is increased. With this system architecture, the hardware such as DLC, PIM, Power Supply etc. is reduced and easy moves, adds, and changes can be realized.

Standard TDM Hardware	Peer to Peer IP Hardware
▪ Line & Trunk Cards ▪ Application Processors ▪ Firmware Processors	▪ SPN-8IPLA IP PAD ▪ PZ-M606-A ▪ PN-24IPLA IP PAD* ** The PN-24IPLA is a daughter board for the 8IPLA when up to 32 IP PADs for desired.*

Enhanced Built-in Firmware Processor (FP) on MP

The Firmware Processor card (FP) provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, and monitors I/O Bus and PCM BUS. When the system consists of three PIMs or more, one each of this card is mounted respectively in PIM 2, PIM 4, and PIM 6.

Extended Application Processor (AP) Port Capacity

The NEAX 2000 IPS provides maximum 256 AP ports and it is independent of the 512 ports for the Line/Trunk (LT), therefore, more AP cards such as T1/E1 digital link cards can be used in the system.

Universal Slot

One PIM provides 12 card slots for Line/Trunk (LT). Also, these card slots can be used for Application Processor (AP) cards without complicated limitation. This makes easy quotation and installation, and more AP cards can be mounted in one PIM.

Unified Circuit Card Size

All circuit cards for the NEAX 2000 IPS are designed in one size (PN-type), and installed in the PIM. This maximizes the efficiency of slot utilization of the PIM.

High Density Line/Trunk Cards

The major line/trunk cards used in the NEAX 2000 IPS are provided with 8 circuits per card. This allows the physical system size to be compact.

DC/DC Power Supply for –48V

The PIM houses optional DC/DC Power Supply for the cards which require –48V power such as the CSI card used for interface of Zone Transceiver of wireless system. Since this power supply is mounted in the space under the AC/DC power, no additional Power Module/card slots are required.

Built-in DRS (Device Registration Server) on MP

The NEAX 2000 IPS incorporates DRS (Device Registration Server) on the MP. DRS provide Log-in/Log-out management of DtermIP including Registration and Authentication. Also, the built-in DRS can be inter-worked with DHCP server to provide easy administration on IP address.

Office Data Backup Enhancement

The office data of the NEAX 2000 IPS is stored in Flash ROM; therefore the backup period is extended compared with previous IVS series which were using RAM with battery.

Various Installation Methods

To meet the specific needs of the customer’s environment, the NEAX 2000 IPS provides the following installation methods:

- Floor Standing Installation
- Wall-mounting Installation
- IEC standard 19 inch Rack-mounting Installation

Station to Station Connection

For DtermIP to DtermIP connection (Peer to Peer connection), the voice data is transmitted and received directly between DtermIPs on the LAN. For Dterm Legacy terminal connection, the IP-PAD card and VCT card are required to transmit and receive the voice data. These cards are used to control and convert the voice data. The MP card in either of the connections above manages the control signals.

CCIS Connection

DtermIP to DtermIP connection (Peer to Peer connection) via CCIS is available only when the destination office is NEAX 2000 IPS or NEAX 2400 IPX. The system provides only Point to Multipoint connection.

Maintenance

MATWorX IPS is used as the maintenance program for the NEAX 2000 IPS. Direct connection (RS-232C), Modem connection and LAN (TCP/IP) connections are available to connect to the MAT (Maintenance Administration Terminal).

Dual MP System

The system complies with dual control system on Main Processor.

Note: *Since the system employs Cold Standby processing in MP changeover, the calls in progress are terminated as a result of the MP changeover. Also, during the MP changeover, the call originating/receiving and service feature access are not effective. (It takes about 30 to 60 seconds to complete the MP changeover.)*

Remote PIM over IP with Survivability

The NEAX 2000 IPS can have a PIM installed at a remote site through an IP network. At the main site, the NEAX 2000 IPS/NEAX IPS^{DM} is installed and NEAX 2000 IPS/NEAX IPS^{DM}/ NEAX IPS^{DMR} are installed at the remote site. The main site controls call processing and service feature access for station users located at both the main and remote sites. When the Remote PIM cannot be connected with main site due to the IP network and/or main PBX failure, the Remote PIM initializes the system and re-starts operation by its own Main Processor (survival mode). In the survival mode, almost all service features are provided to the station users accommodated in Remote PIM. When the IP network/main PBX recovers, the Remote PIM can be restored to normal mode with a system initialization by manual operation or automatically (Selectable by system data setting).

- IPS^{DM} with CP24-A MP
- IPS^{DMR} with CP31-A MP

Software Architecture

Generic Program

Description	Remarks
64 Port Sys Software (FD)	Basic Business/Hotel/Motel Features for: 64 LT Ports, 5 T1's /E1's, 5 ISDN-PRI DCH's, 48 ISDN-BRI Trunks.
Note: *The MP (PN-CP24) comes with 48 Ports of basic software, which supports up to 48 LT Ports & 1 T1.*	
Description	**Remarks**
Key Keeper (FD)	Floppy Disk that holds selected Key Files for Capacity Options
Capacity options (used w/Key Keeper)	
LT 64 Ports	64 Port Line/Trunk Key (incremental) Software and provides LT Port Licenses in 64 port increments from 64 to1020 ports. Stand alone system maximum 512 LT ports, Remote PIM Network maximum 1020 LT ports.
CCIS Link (1)	Adds support for one CCIS Link
CCIS Link (4)	Adds support for four CCIS Links
CCIS Link (8)	Adds support for eight CCIS Links
IPT Card (1)	Adds support for PTP CCIS or one IP Trunk Card
IPT Card (4)	Adds support for up to four IP Trunk Cards
IPT Card (8)	Adds support for up to eight IP Trunk Cards
ECCIS	Adds Event Based CCIS capability (used w/CCIS keys)
Wireless	Adds Wireless Capability Supports 128 ZT's & 256 PS's
Wireless – 8 PS License	Adds additional licenses (increments of 8) for over 256 PSs
T1/E1 6 to 10 Cards	Expands T1/E1 Capacity between 144 to 240 Channels.
ISDN DCH 5 to 8 Cards	Expands ISDN PRI Capacity between 5 DCH Cards and 8 DCH Cards.
R-PIM 1 Site License	Site License for DMR/IP Remote PIM. One License required per Remote Site.
IPT Card (1)	Adds support for PTP CCIS or one IP Trunk Card
8 Seat License	Required for eight or less Dterm IP Terminals and IP Soft-Phones. Incremental eight Seat License for Peer to Peer IP ***Note:*** *IP Seat Licenses are Accumulative. (i.e. 24 IP = three 8 Seat License)*
Soft-Phone 4 Seat License	4 seats Dterm SP20 licenses - Requires an available 8 SEAT LICENSE seat per Soft-phone seat activated. **Note:** Soft-Phone 4 *Seat Licenses are used in addition with 8 Seat License when using the SP20 SoftPhone*
SP30 - 4 Seat License	4 seat Dterm SP30 license. Incremental one seat used for each Soft-Phone client. **Note:** Soft-Phone 4 *Seat Licenses are used in addition with 8 Seat License when using the SP20 SoftPhone*

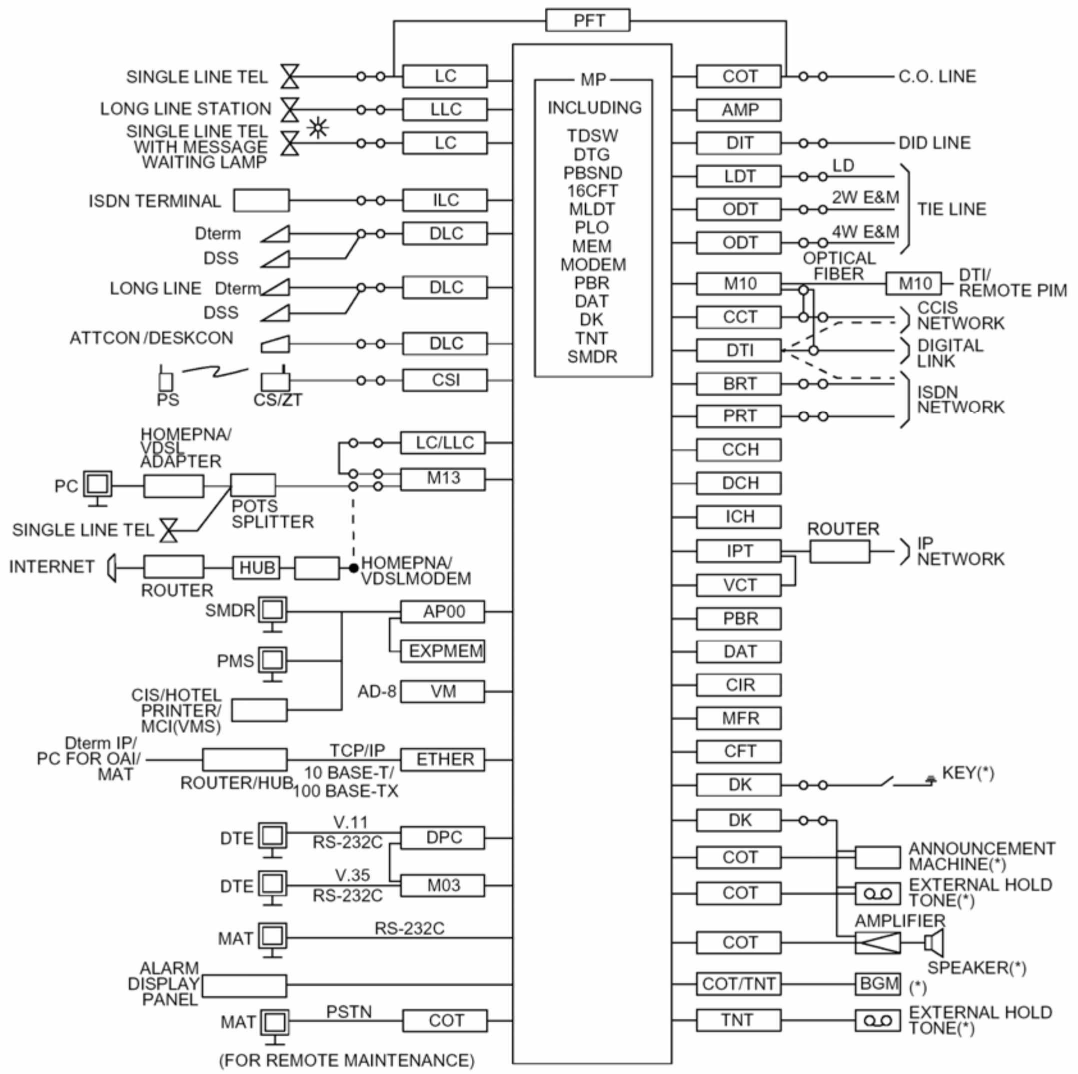
Technical Terms

SYMBOL	DESCRIPTION	SYMBOL	DESCRIPTION
AP00	SMDR/Hotel Application Card	LC	Line Circuit Card (for Single Line Telephone)
AP01	OAI Interface Card	LDT	LD Trunk Card
AUC	Analog Universal Circuit Card (Long Line Circuit, DID Trunk)	M03	V.35 DTE Interface Card
BGM	External Music Source for D^{term} Back Ground Music Service	M10	Optical Interface Card
BRT	Basic Rate Interface Trunk Card	MAT	Maintenance Administration Terminal
CCH	Common Channel Handler Card	MDF	Main Distribution Frame
CFT	6/10 Party Conference Trunk Card	MEM	Main Memory
CIS	Call Information System	MFR	MF Receiver/ MFC Receiver/Sender Card
CIR	CALLER ID Receiver Trunk Card	MLDT	Melody Trunk
COT	C.O. Trunk Card	MODEM	Modem
CSI	CS/ZT Interface Card	MP	Main Processor Card
CS/ZT	Cell Station (For Australia/Others) Zone Transceiver (For North America/ Latin America)	PFT	Power Failure Transfer
DAT	Digital Announcement Trunk Card	PMS	Property Management System
DCH	D-channel Handler Card	OAI	Open Application Interface
DIT	DID Trunk Card	ODT	OD Trunk Card (2/4 wire E&M)
DK	External Relay/Key Interface Card	PBR	PB Receiver Card
DLC	Digital Line Circuit Card (for D^{term}, ATTCON, DESKCON)	PBSND	PB Sender
DPC	Data Port Controller Card	PLO	Phase Locked Oscillator
DSS	DSS Console	PS	Personal Station
DTE	Data Terminal Equipment	PRT	ISDN Primary Rate Interface Trunk Card
DTI	Digital Trunk Interface Card	SMDR	Station Message Detail Recording
DTG	Digital Tone Generator	TDSW	Time Division Switch
ETHER	Ethernet Control Card	TNT	Tone/Music Source Interface Card
EXPMEM	Memory Expansion Card	VCT	CODEC Card
ICH	ISDN-channel Handler Card	VM	Voice Mail Card
ILC	ISDN Line Circuit Card	16CFT	16 Circuit Four Party Conference Trunk
IPT	IP Trunk Card	KEY	External Key

Trunking Diagram

This figure shows a typical trunking diagram of the NEAX2000 IPS system.

NOTE: *The equipment marked with (*) is provided by the customer.*



NEAX® 2000 IPS^{DM}

The NEAX IPS^{DM} (Internet Protocol Server Distributed Model) is equipped with all the features and functions of the NEAX 2000 IPS, with a smaller space requirement. It is a full-featured PBX that supports advanced networking, pure peer-to-peer IP telephony connectivity and traditional TDM switching capabilities. Designed primarily for pure converged IP networks, the NEAX IPS^{DM} can also accommodate a mixed (i.e., TDM and IP) converged IP network or standalone solution.

The NEAX IPS^{DM} supports up to 956 peer-to-peer IP stations and 40 TDM ports in a single modular chassis. Up to three chassis can be stacked providing maximum capacity of 120 legacy TDM ports while still supporting as many as 828 peer-to-peer IP stations or more depending on the amount of TDM stations used. It uses the same CPU, line/trunk cards, application processor cards and software of the NEAX 2000 IPS and comes equipped for 19" rack mounting. It offers superior port density; each chassis only occupies two Rack Units (2RU).

Characteristics of the NEAX IPS^{DM}

Compact and Small Size MODULAR CHASSIS

One MODULAR CHASSIS provides 6 card slots /40LT ports and up to 3 MODULAR CHASSIS can be used per system. (24 virtual LT ports are available per MODULAR CHASSIS in addition to 40LT ports.)

2 types of MP (Main Processor)

MP can be selected from the following options by customer requirements.

- PN-CP24-A for IPS^{DM}, the same MP as the NEAX 2000 IPS.
- PN-CP31-A for IPS^{DMR} , the following functions are removed from the CP31: DAT / DK00 / 1 RS232C Port for MAT / MN Alarm Indication

Power Failure Transfer (PFT)

Power Failure Transfer (PFT) for the IPS^{DM} is provided with PZ-4PFTA card. The PZ-8PFTB for the NEAX 2000 IPS is not available for the IPS^{DM}.

IPS^{DM} Installation Methods

Wall Mount Installation is not available. The NEAX IPS^{DM} can be installed on the desktop or into the 19-inch rack.

NEAX® 2000 IPS^{DMR}

The NEAX IPS^{DMR} (Internet Protocol Server Distributed Model Remote) is a NEAX IPS^{DM} that has been optimized for Remote PIM over IP applications. The NEAX IPS^{DMR} uses the SPN-CP31 as the Main Processor. The SPN-CP31 is a cost down CPU to compete with Mitel 3100, NBX25, and CISCO 2600 Series. This system targets users who have up to 15 relatively small offices that accommodate 10-30 extensions at the Remote Site.

The MP card at Remote Site has the same system data as that at Main Site, because Remote Site automatically gets the data from Main Site at the time of setup. In normal operation, Main Site automatically copies the system data to Remote Site through the network once a day.

Because the CP31 is a cost down CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in modem.
- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

System Outline

- The MP card at Main Site controls system processing, and Remote Site follows the Main Site.
- Remote Site can accommodate most terminals and trunks such as Dterm, Single-Line telephone, PS, DtermIP, COT, ISDN, etc. The Attendant Console, Dterm Attendant position, Add-on Module and DSS/BLF are **not** supported at the Remote Site.
- Local Switch (TDSW) at Remote Site controls connections within the Remote Site if possible.
- In the case of connections between Main-Remote and Remote-Remote, the voice path is connected via Peer-to-Peer or IP-PAD.
- If the communications between Main-Remote are interrupted, the Remote Site survives by itself after the system reset.

Advantages

- The system regards the terminals accommodated in both Main Site and Remote Site as the extensions in the same office. Therefore, the service transparency is superior to CCIS.
- Remote PIM over IP has no limitation of distance between Main and Remote.
- Remote Site has a switching function at local. This provides the effective configuration of C.O. line. In addition, the Remote Site can accommodate AP cards. This is an advantage to accommodate ISDN lines especially.
- The Remote Site survives by itself even if the link between Main and Remote is disconnected. Therefore, the impact to users at Remote Site will be smaller if the link between Main and Remote is disconnected.
- This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than DtermIP at remote location or the Media Converter (MC) accommodation.

Remote PIM over IP

Remote PIM over IP targets users who have 1-15 relatively small offices that accommodate 10-30 extensions at the Remote Site. When IPS^{DMR} and 2000 IPS PIM are installed at remote site, and connected to a 2000 IPS or IPS^{DM} at main site over IP network, the Main Site system controls and maintains the remote DM and PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

IPS^{DMR}: IPS Distributed Model Remote (with CP31-A)
IPS^{DM}: IPS Distributed Model (with CP24-A/B)

The NEAX IPS-DMR is designed primarily for distributed IP networking but also supports traditional analog and digital trunks for connection to the Public Switched Telephone Network (PSTN). The NEAX IPS-DMR supports up to 128 peer-to-peer IP stations and 40 TDM ports in a single modular chassis. Up to two chassis can be stacked providing maximum capacity of 80 TDM ports while still supporting as many as 128 peer-to-peer IP stations.

Note: *The MP card at Remote Site has the same system data as the CPU at the Host Site; the Host Site automatically downloads system data to the Remote Site at the time of setup. In normal operation, Main Site automatically downloads a copy the system data to Remote Site through the network once a day.*

Because the CP31 is designed as a Remote PIM CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in modem.
- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

Network Conditions and Payload

Item	Requirement	Remarks
Protocol	TCP/IP transparent	
Maximum Delay Time	120ms(one way)/240ms(return) 150ms(one way)/300ms(return)	Support the quality class A, B of IP Telephone

Bandwidth Requirement

Established Voice Calls		With G7.23.1 (5.3k/6.3k) Compression	With G729a (8k) Compression	Without Compression (G.711)
6	Control	4.1 Kbps	4.1 Kbps	4.1 Kbps
	Voice	31.8/37.8 Kbps	48 Kbps	432 Kbps
8	Control	4.3 Kbps	4.3 Kbps	4.3 Kbps
	Voice	42.4/50.4 Kbps	64 Kbps	576 Kbps
12	Control	4.3 Kbps	4.3 Kbps	4.3 Kbps
	Voice	63.6/75.6 Kbps	96 Kbps	864 Kbps
16	Control	4.5 Kbps	4.5 Kbps	4.5 Kbps
	Voice	84.8/100.8 Kbps	128 Kbps	1152 Kbps
24	Control	4.5 Kbps	4.5 Kbps	4.5 Kbps
	Voice	127.2/151.2 Kbps	192 Kbps	1728 Kbps
32	Control	4.9 Kbps	4.9 Kbps	4.9 Kbps
	Voice	169.6/201.6 Kbps	256 Kbps	2304 Kbps
48	Control	4.9 Kbps	4.9 Kbps	4.9 Kbps
	Voice	254.4/302.4 Kbps	384 Kbps	3456 Kbps
64	Control	5.8 Kbps	5.8 Kbps	5.8 Kbps
	Voice	339.2/403.2 Kbps	512 Kbps	4608 Kbps
72	Control	5.8 Kbps	5.8 Kbps	5.8Kbps
	Voice	381.6/453.6 Kbps	576 Kbps	5184 Kbps
96	Control	6.7 Kbps	6.7 Kbps	6.7 Kbps
	Voice	508.8/604.8 Kbps	768 Kbps	6912 Kbps

Note: *This information is an estimation based on an established call. Slightly Higher Control values will occur at time of call origination and termination.*

Base values

- Originating from a station: 9.6 Kbps/Call (estimated)
- Terminating to a station: 5.76 Kbps /Call (estimated)
- Originating to C.O: 11.5 Kbps/Call (estimated)
- Terminating from C.O: 5.76 Kbps/Call (estimated)
- Keep Alive to Remote Site: 0.032Kbps (estimated)
- Other control packets for Remote Site: 4Kbps (estimated)
- G.723.1 voice: 5.3Kbps (one-way)
- G.729a voice: 8Kbps (one-way)
- G.711 voice: 64Kbps (one-way)

The above base values are primarily used for call setup with the exception of keep alive; 0.032Kbps with no voice traffic. Connections between IP PAD are half duplex, established call utilization is G.711 voice: 64Kbps, G.723.1 voice: 5.3/6.3Kbps, or G729a voice: 8Kbps. Peer-to-Peer IP station calls are full duplex, compression can be specified by location numbers in system data. Peer-to Peer IP station calls even though full duplex will utilize one-way for Bi-directional networks such as T1. Peer-to Peer IP station calls over Asymmetrical networks such as ADSL may realize higher bandwidth utilization, compression can be specified by location numbers in system data.

Advantages

The system regards the terminals accommodated in both Host Site and Remote Site as the extensions in the same office. Feature transparency is superior to CCIS.

The Digital Remote PIM cannot accommodate AP cards; Remote PIM over IP can accommodate AP cards such as ISDN PRI and T1.

This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than Dterm IP at remote locations.

Since all Remote PIM over IP sites are treated as extensions in the same office, software and applications only have to be implemented in the host site. This provides centralized use of application for example distributing ACD agents in the DMR locations. CCIS requires each location to have separate software and applications.

CCIS over IP can be combined with Remote PIM over IP to accommodate larger network configurations. Up to 255 host sites can be connected via CCIS, each host site can have up to 15 Remote PIM over IP locations.

Service Conditions

1. Host site can be NEAX 2000 IPS, NEAX IPS DM, or NEAX 2000 Retrofit system. Remote PIM over IP is available in any combination of the following CPUs.
   Main Site: CP24-A/B, CP27-A, CP26-A, CP28-A
   Remote Site: CP31-A, CP24-A/B, CP27-A, CP26-A, CP28-A

2. Software and Key FD for the whole system must be loaded at the Host Site. No software or key’s can be loaded into the Remote Site.

3. All system data changes for the whole system must be performed in the Host Site. No system data changes can be done in the Remote Site.

4. The CPU card at Remote Site has the same system data as the CPU at Main Site; the Host Site automatically downloads its system data to the Remote Site at the time of setup. In normal operation, Host Site automatically copies the system data to Remote Site through the network once a day.

5. Remote Site automatically operates by itself (survival mode) when Keep Alive signal (sent every 30 sec) between the Host Site and Remote Sits is interrupted. When Keep Alive is interrupted the Remote Site is reset to change the operation from normal mode to survival mode.

6. Remote Site in survival mode checks at 30 seconds intervals if the communications to Main Site are possible. When Keep Alive is detected, the Remote Site automatically is reset to change the operation from survival mode to normal mode.

7. When unstable conditions occur in the network, the Remote Site can be manually set to survivable mode (override automatic) until stability in the network is established. This prevents the Remote Site from resetting normal mode to survivable mode etc.

Required Hardware and Software

Host Site

Equipment Name	Remarks
PZ-M606-A	On board Ethernet Interface card
SPN-8IPLA IP PAD	8 Port PAD with built-in compression
PZ-24IPLA	24 Port PAD Expansion, mounts on SPN-8IPLA
R-PIM 1 Site License	1 required for each Remote site

Note: *Registration of Host CPU and software required*

DMR Site

Equipment Name	Remarks
PZ-M606-A	On board Ethernet Interface card
SPN-8IPLA IP PAD	8 Port PAD with built-in compression
PZ-24IPLA	24 Port PAD Expansion, mounts on SPN-8IPLA

Note: *Registration "not" required*

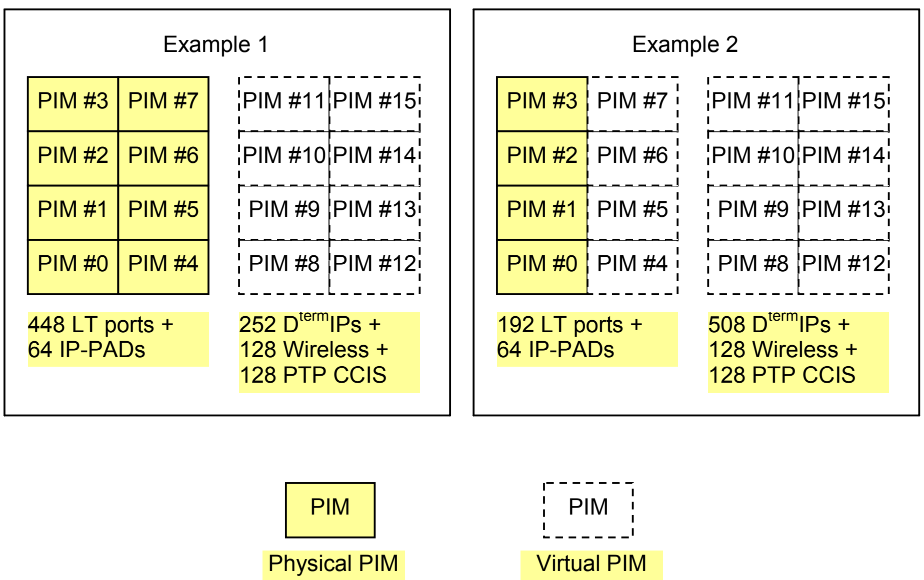
Chapter 2 System Configuration

Module Configuration

The NEAX® 2000 IPS consists of single or multiple Port Interface Modules (PIM) depending on the system configuration, and there are two types of PIMs; "Physical" PIM and "Virtual" PIM. The Physical PIM is "hardware" PIM which is used to accommodate an MP, FPs, IP PADs, legacy LT cards, AP cards, and power supply units. One Physical PIM provides up to 64 LT ports and up to 8 Physical PIMs can be accommodated in a Stand Alone system. The Virtual PIM is a "software" PIM and provides up to 64 ports per PIM for use by system programming as DtermIP telephones, Wireless PS stations or Peer to Peer (PTP) CCIS trunks. The system consists of up to 16 PIMs, by the combination of Physical PIMs and Virtual PIMs, thus providing 1020 ports. When the use of Virtual PIMs exceeds 8 then the number Physical PIMs is reduced by one for each additional Virtual PIM required.

The illustration below shows examples of 1020-port configuration by the combination of TDM LT ports, Dterm IP telephones, Wireless PS stations and Peer to Peer (PTP) CCIS trunks.

Figure 2-1 System Configuration with Dterm IP (1020-Port Configuration)



The figure 2-2 shows another example of 1020-port configuration by combination of legacy LT ports and Dterm IP telephones.

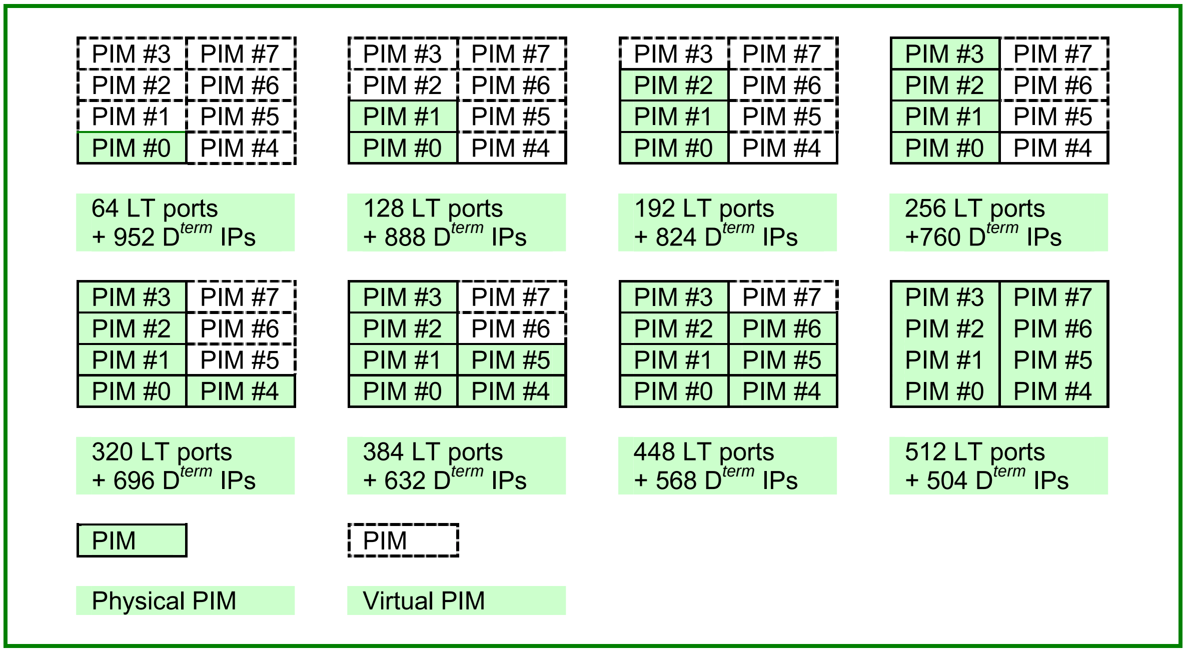


Figure 2-2 System Configuration with Dterm IP (1020-Port Configuration)

Installation Methods

The NEAX2000 IPS provides three installation methods as follows:

- Floor Standing Installation
- Wall Mounting Installation
- 19-inch Rack Mounting Installation

Floor Standing Installation

In Floor Standing Installation, the NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).

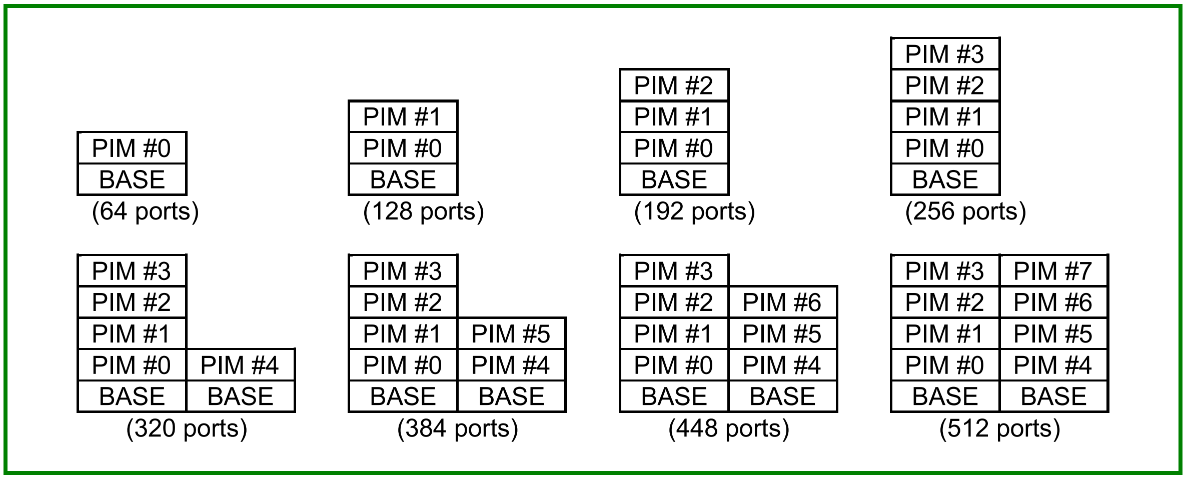


Figure 2-2 System Configuration in Floor-standing Installation

Wall-mounting Installation

The NEAX 2000 IPS can be wall-mounted with single or multiple PIM configurations (maximum of eight PIMs).

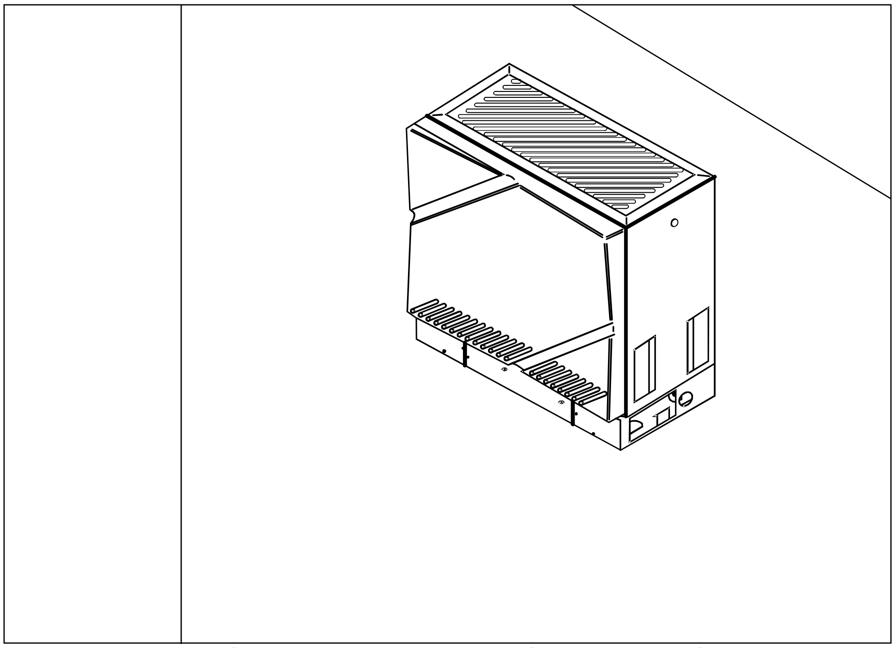


Figure 2-3 Wall-mounting Installation

19 inch Rack-mounting Installation

The NEAX 2000 IPS can be mounted in the IEC-standard 19 inch rack up to four PIMs.
(IEC: International Electro-technical Commission)

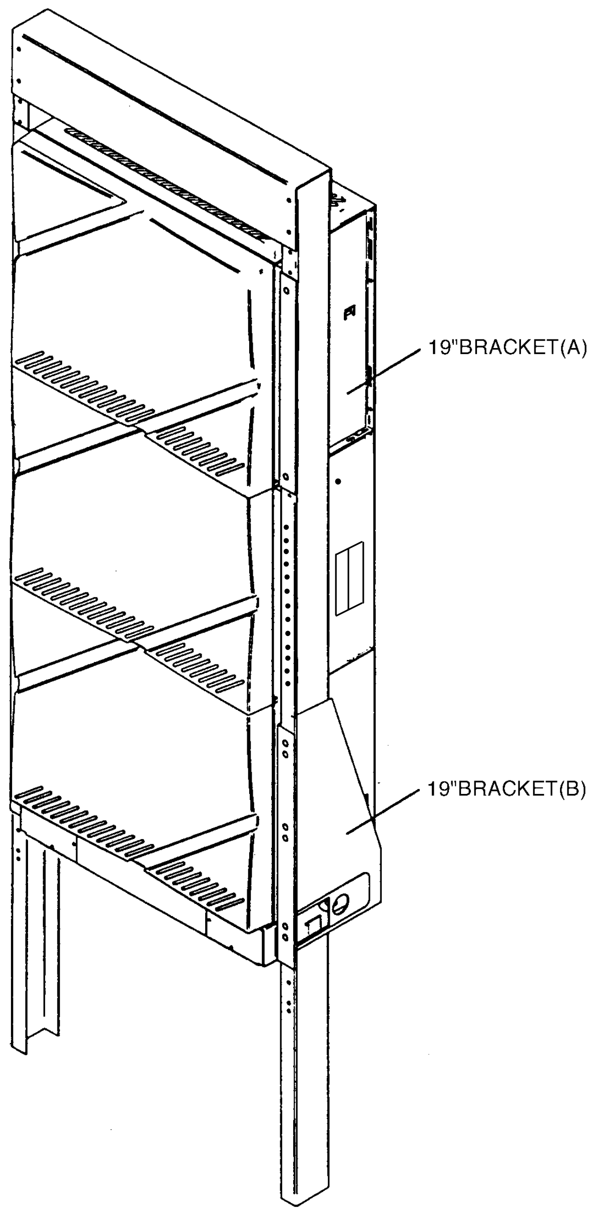


Figure 2-4 19-inch Rack-mounting Installation

Modules and Installation Hardware

The NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).

Modules

(1) **Port Interface Module (PIM)**

A PIM provides 13 card slots for common control, Line/Trunk (LT), and Application Processor (AP) cards. It also houses an AC/DC Power Supply, DC/DC Power Supply (for -48V), and batteries for protection from short-term (about 30 min.) power interruption. Four champ connectors for Line/Trunk (LTC 0 to 3) are located at the lower front side of the PIM. A PIM provides a maximum of 12 card slots for Line/Trunk (LT) and Application Processor (AP) cards. At maximum configuration, the system is comprised of 8 PIMs.

There are two types of PIM (PIMMD and PIMMF) depending on the system type as follows.

Type of PIM	Single MP System	Dual MP System
PIM MD	Used for PIM 0-7	Used for PIM 1-7
PIM MF	Not used	Used for PIM 0

PIM MD (PIM3)	PIM MD (PIM7)
PIM MD (PIM2)	PIM MD (PIM6)
PIM MD (PIM1)	PIM MD (PIM5)
PIM MD (PIM0)	PIM MD (PIM4)
(Single MP System)	

PIM MD (PIM3)	PIM MD (PIM7)
PIM MD (PIM2)	PIM MD (PIM6)
PIM MD (PIM1)	PIM MD (PIM5)
PIM MF (PIM0)	PIM MD (PIM4)
(Dual MP System)	

Unit Configuration

(2) **Battery Module (BATTM)**

The BATTM is an optional module for installing optional long-term (about 3 hours) backup batteries. The BATTM is designed to accommodate batteries covering up to a 4-PIM system (2 BATTMs support maximum system configuration). The BATTM is available for Floor Standing Installation. (When the system is Wall-mounting/19 inch Rack-mounting configuration, the BATTM cannot be installed with the PIM.)

Modules

Abbrev	Description	Remarks
PIMMD	SN1617 PIMMD	Single MP System: PIM 0 - PIM 7 Dual MP System: PIM 1 – PIM 7
PIMMF	SN1658 PIMMF	Single MP System: Not used Dual MP System: PIM 0
BATTM	SN1619 BATTMB	1 per STACK, Max.2 per system

Installation Hardware

Base/Top Assembly

The Base/Top Assembly includes a Base Unit and a Top Cover for the PIM. One Base/Top Assembly is required for each PIM stack. The Base Unit also serves as the AC power distribution panel for up to a four PIM configuration.

Hanger Assembly

The Hanger Assembly is used for Wall-mounting Installation. One set of Hanger Assembly is required for each PIM.

19 inch Bracket

The 19-inch Bracket is a set of hardware used for 19-inch Rack-mounting Installation. The 19-INCH RACK BRACKET (A) is installed on both sides of the PIM. One set of 19 inch Bracket (A) is required for each PIM. The 19-INCH RACK BRACKET (B) is installed at the BASE of stack. One 19-INCH BRACKET (B) is required for each stack.

If the system is 2 PIM or more configurations with 19-INCH BRACKET (B), one set of 19-INCH BRACKET (A) is also required for the topmost PIM.

Optional Brackets

The Mounting Bracket is used for Floor Standing Installation. Without Mounting Bracket, 1.1G shockproof is provided for 1 to 3-module stack and 0.5G shockproof is provided for 4 or more module stack. To enhance the shockproof capability to 1.1G, one set of Mounting Bracket is required for each 4 or more module stack and attached to the topmost PIM.

The I/F Bracket is used for Floor Standing Installation to joint the neighboring topmost PIM in 6 PIM or more configurations. One set of I/F Bracket is required for multiple stacks.

The Base Tray Assembly is used for Floor Standing Installation for stationary equipment (UL complied). One set of Base Tray Assembly is required for each stack.

Installation Hardware

Abbrev	Description	Quantity
Top Cover	TOP COVER ASSEM	1/STACK (BASE ASSEM is local supply)
Base/Top ASSEM	SN1545 BASERE	1/STACK
Hanger Assem	HANGER ASSEM (UL)	1/PIM (Wall-mounting Installation)
19 inch Bracket	19 INCH RACK BRACKET (A)	1/PIM (19 inch Rack-mounting Installation)
	19 INCH RACK BRACKET (B)	1/STACK (19 inch Rack-mounting Installation)
Mounting Bracket	MOUNTING BRACKET	OPTION (1/STACK)
I/F Bracket	I/F BRACKET ASSEM	OPTION (1/SYSTEM)
Base Tray	BASE TRAY ASSEM	OPTION (1/STACK)

NEAX 2000 IPS SYSTEM POWER SUPPLY

AC/DC Power Supply

The AC/DC Power Card is mounted in the left side of each PIM. The AC/DC Power card provides power to all circuit cards, which reside in the PIM. AC power requirements are as follows:
Input Voltage: 90 to 132 Vrms or 180 to 264 Vrms (selectable by switch) 50/60 Hz

DC/DC Power Unit

The DC/DC Power Unit is mounted under the AC/DC Power Card and generates -48 V power for the circuit cards that need such power.

Battery Backup

Internal Short-term option

For customers requiring battery backup, short-term and/or long-term options are available. Two 3.4AH batteries are required per PIM, and installed inside of each PIM. Backup time is approx. 30 minutes when PHS (Wireless PS) is not accommodated and approx. 10 minutes when PHS (Wireless PS) is accommodated in the system.

External Long-term option

Two 24AH batteries are required per each 2 PIMs, and installed inside of Battery Module in a stack basis. Backup time is approx. 3 hours when PHS (Wireless PS) is not accommodated and approx. 2 hours when PHS (Wireless PS) is accommodated in the system. The batteries are varied depending on the requested backup time. The battery shall be locally provided.

Circuit Cards

The circuit cards used for NEAX 2000 IPS are divided into the following three types. According to these card types, the mounting locations of card and port allocation of the Time Division Switch are varied.

- Common Control Cards
  - Main Processor (MP)
  - Firmware Processor (FP)
  - Ethernet
  - Power

- Line/Trunk (LT) Cards
  - IP PAD, Line Circuit (LC), Central Office Trunk (COT), Tie Line Trunk (LDT/ODT), etc.

- Application Processor (AP) Cards
  - SMDR/PMS/CIS/Hotel Printer Interface (AP00)
  - T1/E1 Digital Trunk Interface (DTI)

IPS System Conditions

2000 IPS is an IP communication system that integrates voice terminals through Peer-to-Peer connection to the IP network. The system is a hybrid system to accommodate both IP multiline terminals (DtermIP) and the Legacy PBX's terminals (Legacy terminal). Line/Trunk cards and Application Processor cards can be mounted in the system to provide the Legacy PBX features that use the Time Division Switch (TDSW).

Station-To-Station Connection

Station-to-Station connection is available on the LAN. For DtermIP-to-DtermIP connection (Peer-to-Peer connection), the voice data is transmitted and received directly between DtermIPs on the LAN. For DtermIP-to-Legacy terminal connection, the IP-PAD card is required to transmit and receive the voice data. This card is used to control and convert the voice data. The MP card manages control signals in both types of connections.

Public Network/TIE Network Connection

The system can be connected with a Public Network or Tie Line Network. When the DtermIP communicates with the DtermIP/Legacy terminal in the destination office via Public Network or Tie Line Network, the IP-PAD card and the trunk card are required to transmit and receive the voice data.

CCIS Connection

The system can be connected with the IP network by No. 7 Common Channel Inter-office Signaling (CCIS) via the Virtual IPT, when the destination office is 2000 IPS or 2400 IPX. For DtermIP-to-DtermIP connection via CCIS (Peer-to-Peer connection), the voice data is transmitted and received directly between DtermIPs via the IP network (CCIS via IP). For DtermIP-to-Legacy terminal connection via CCIS, the IP-PAD card is required to transmit and receive the voice data. This card is used to control and convert the voice data. The MP card has a built-in Virtual IPT and the Virtual IPT manages control signals in both types of connections.

H.323 Connection

The system can be connected with the IP network by ITU-T recommendation H.323 protocol. The system can be connected to the terminal and network equipment according to H.323 protocol. For DtermIP-to-DtermIP connection via the IP network with H.323 protocol, the IPT card and IP-PAD card are required to transmit and receive the control signal and voice data. For voice compression, the 4VCT card is required. For Legacy terminal connection via the IP network with H.323 protocol, the IPT card is required. For voice compression, the 4VCT card is required.

Conditions for Overall System

- To connect the MP (PN-CP24-A/PN-CP24-B/PN-CP27-A) card to the LAN, ETHER (PZ-M606-A) card is required on the MP card.
- When you set or change the system data, the system data backup must be executed by CMEC Y=6>0: 0. If the system is turned off or MP card is reset without the backup, the data that has been set or changed is cleared.
- System data can be saved to the flash memory on the MP card on a daily basis. The data setting to execute the regular system data backup is required.
- While saving the system data to the flash memory, the “SYSD” lamp on the MP card flashes. Do not turn off or reset the system while the “SYSD” lamp is flashing.
- After executing the system data memory all clear, FP No. 00 is set to MP built-in FP and FP No. 01-03 are set to Signaling Converter (Virtual FP) in default.
- One Virtual FP/AP card provides 64 ports to connect the Line/Trunk cards.
- The DTMF sender signal width of Dterm/DtermIP is 112-128 ms.
- When upgrading the software of the system from Series 3200 R6.2 or before to Series 3300 or later, the office data conversion is required.
- When connecting the MP card/IP-PAD card/IP terminal and the switching HUB which Spanning Tree (IEEE 802.1d) is available, communication failures shown below may occur. Set up the Spanning Tree invalid by the switching HUB.

  – IP terminal fails to connect to 2000 IPS.
  – IP terminal cannot communicate with the IP terminal.
  – IP terminal cannot communicate with the SLT/Dterm.
  – Remote Site cannot change over to the normal mode in the Remote PIM over IP system.
- When connecting the MP card/IP-PAD card/IP terminal and the switching HUB which LACP (Link Aggregation Control Protocol: IEEE 802.3ad) function is available, communication failures shown below may occur. Set up the LACP function invalid by the switching HUB.

  – Remote Site cannot change over to the normal mode in the Remote PIM over IP system.

Conditions for Dterm IP

- For the DtermIP, an AC-DC adapter or inline power patch panel is required.
- The DtermIP cannot be accommodated in the TDM based Remote PIM.
- The hold tone for DtermIP is only “Minuet”. The hold tone set by CM48 Y=3 are not effective for DtermIP.

Conditions for Public Network/TIE Line Network Connection

- For the DtermIP communication between offices, the IP-PAD card is required.
- Peer-to-Peer connection is not available in this connection.

Conditions for Peer-to-Peer Connection

- For the communication between DtermIPs, the voice data is transmitted and received directly, without converting voice packets into PCM and voice compression in the system.

Conditions for CCIS Connection

- Peer-to-Peer connection between DtermIPs via CCIS is available only when the destination office is 2000 IPS or 2400 IPX.
- The Virtual IPT can be connected to a maximum of 127 trunks.
- The Virtual IPT provides only Point-to-Multipoint connection.
- When a call over Peer-to-Peer connection via CCIS is put on hold and then answered at the same station, Elapsed Time Display returns to 0:00:00.
- When the destination office uses the physical IPT card, for example, when connecting to the former PBX system, the IPT card and 4VCT card are required in both offices.
- Conditions for Link Down Notice for CCIS connection are shown below.
  - Link Down Notice is available only for Dterm and DtermIP accommodated in the 2000 IPS and IPSDM/IPSDMR. This is not available for a single line telephone and Attendant Console.
  - For message display, Dterm/DtermIP with 24-digit or more LCD is recommended. 16-digit LCD may not display all messages properly.
  - Notification message can be displayed regardless of idle or busy state of Dterm/DtermIP, writing the message over the present display. After six seconds, the display returns to the time display automatically.
  - The system detects a Link Down on the condition that TCP connection between offices is interrupted. The Link Down is notified to the Dterm/DtermIP at 15-20 seconds later from the system detects the Link Down.
  - Link Down Notice is available only for the CCIS connection via Virtual IPT. CCIS connection with CCT/DTI card or LDT/ODT card is not available.
  - When the link between offices connected by CCIS via Virtual IPT is interrupted, the lamp of Dterm/DtermIP button becomes the state as shown below. Then press the button, the LCD of the Dterm/DtermIP displays the following.

COLOR AND STATE OF BUTTON		STATE AND OPERATION	LCD DISPLAY
Red/Flashing (Momentarily)	0.125 seconds ON-0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON-0.5 seconds OFF	Press the button after Link Down occurrence	Link Down to CCIS
OFF	-	Link restoration	-
OFF	-	Press the button after Link restoration	Normal Condition: CCIS

  - When the link between offices recovers, the flashing lamp of the button goes out.

Conditions for DRS

DRS=Device Registration Server

- The System-based DRS executes DtermIP registration.
- The Network-based DRS is not available for the DtermIP registration.

Common Conditions for CCIS Connection and H.323 Connection

- The service features requiring continuous voice transmission such as Background Music should not be used because the traffic may reduce overall performance of the LAN.
- In the voice communication via the Internet, the connection and communication delay may occur and the voice quality may deteriorate.
- The Virtual IPT and IPT card does not support Dynamic Host Configuration Protocol (DHCP) service.

Conditions for H.323 Connection

- When connecting to the IP network with H.323 protocol, the IPT card and 4VCT card are required.
- When connecting DtermIP to the IP network with H.323 protocol, the IP-PAD card is required.
- Peer-to-Peer connection between offices is not available.
- Connection via the Intranet is only supported.
- For Voice over IP (H.323), a H.323 Gatekeeper is required for converting between IP address and station number.
- Confirmation test is required for connection to H.323 terminals (Gatekeeper, Gateway, IP terminal, etc.) of other venders.
- The supplementary service defined by H.450 standard is not supported.
- The IP trunk provides only Point-to-Point connection.

Conditions for Maintenance

- MATWorX can be used as the maintenance program for 2000 IPS. Direct connection (RS-232C), Modem connection and LAN (TCP/IP) connection are available to connect to the Maintenance Administration Terminal (MAT).
- When using the MATWorX version 3 or former version, the message (“OK”) for completion of system data memory all clear is not displayed on the MOC window.
- You can check the condition of LAN cable connection by transmitting the ping packet to the ETHER card from PC on the LAN.

Conditions for IP-PAD

- The IP-PAD is required for the following connections.

  – DtermIP-to-Legacy terminal connection
  – DtermIP-to-Public Network/Tie Line Network connection
  – External hold tone connection
  – Conference Trunk (CFT) connection
  – Digital Announcement Trunk (DAT) connection

- The IP-PAD card number depends on the mounting location on the PIM.
- The IP-PAD card uses 32 channels/ports, even if mounting one or no 16VCT card per IP-PAD card.
- Do not pull out the 16VCT card from the PIM or IPTRK BUS CA cable from the IP-PAD/16VCT card, while the MP card is On-line mode. If the 16VCT card or IPTRK BUS CA is pulled out in On-line mode, the IP-PAD card operates abnormally.

- When mounting no 16VCT card, the DTMF signal is not supported to the communication from the IP-PAD to the DtermIP. (In-band (DTMF) tone will be used for the communication between the IPPAD and the DtermIP.)
- When mounting no 16VCT card, the CODEC for IP-PAD is fixed to G.711 and the payload size for IP-PAD is fixed to 40 ms.
- When mounting no 16VCT card and the destination IP terminal does not support the G.711 CODEC, you cannot connect to the terminal via IP-PAD.
- By the office data setting (CM0A Y=73), the used ports for IP-PAD card can be changed to 8/16/24/32 ports. As PN-32IPLA/PN-32IPLA-A (IP-PAD) card requires two physical slots, PN-32IPLA/PN-32IPLA-A (IP-PAD) card cannot be used with 8 ports in 2000 IPS.
- When PZ-24IPLA (24DSP) card is not mounted on the PN-8IPLA (IP-PAD) card or the PN-8IPLA (IP-PAD) card is used with 8/16/24 ports, do not assign the IP-PAD channel No. (CM14) that exceeds the port number of PN-8IPLA (IP-PAD) card or you cannot use it normally.
- PN-8IPLA (IP-PAD) card does not support T.30. Using the PN-8IPLA (IP-PAD) card in 2000 IPS, when the destination office is 2400 IMX, the facsimile transmission is not available. When the destination office is 2000 IPS/IPSDM/IPSDMR (Using PN-32IPLA-A (IP-PAD) card)/2400 IPX, the facsimile transmission by G.711 pass-through is available.
- PN-8IPLA (IP-PAD) card does not support T.30. Using the PN-8IPLA (IP-PAD) card in 2000 IPS, when the destination office is 2400 IMX, the facsimile transmission is not available. When the destination office is 2000 IPS/IPSDM/IPSDMR (Using PN-32IPLA-A (IP-PAD) card)/2400 IPX, the facsimile transmission by G.711/G.726 pass-through are available in the following condition.

  – G.711 pass-through: IP-PAD card (SPN-8IPLA IP PAD-A/SPN-8IPLA IP PAD-B) is mounted.

  – G.726 pass-through: IP-PAD card (SPN-8IPLA IP PAD-B) is mounted.
- PN-32IPLA (IP-PAD) card does not support FAX communication.
- When providing the FAX communication (T.30) with the PN-32IPLA-A card, following condition is required.

  – 16VCT card (SPN-16VCTAA IP PAD-A/SPN-16VCTAA IP PAD-B) is mounted.
- When providing the FAX communication (G.711 pass-through) with the PN-32IPLA-A card, following condition is required.

  – IP-PAD card (SPN-32IPLA IP PAD-D) is mounted.

  – 16VCT card (SPN-16VCTAA IP PAD-B) is mounted.

  (When mounting two 16VCT cards, both 16VCT cards should be SPN-16VCTAA IP PAD-B.)

  – IP-PAD card (SPN-32IPLA IP PAD-E) is mounted.
- When providing the FAX communication (G.726 pass-through) with the PN-32IPLA-A card, following condition is required.

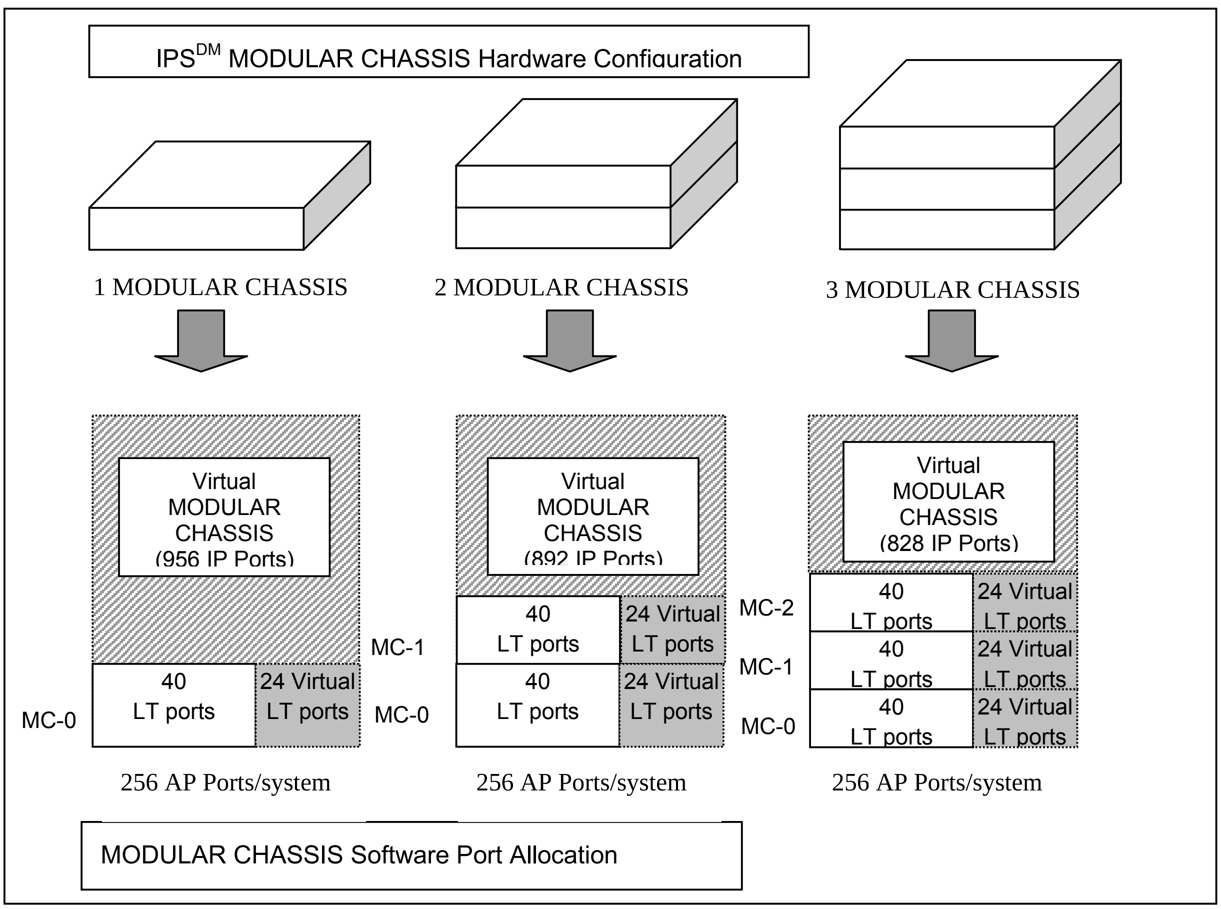
  – 16VCT card (SPN-16VCTAA IP PAD-B) is mounted.

NEAX® IPS^{DM}/IPS^{DMR} System Configuration

NEAX IPS^{DM} Modular Chassis (MC)

The NEAX IPS^{DM} consists of from one to three MODULAR CHASSIS depending on the system configuration. The MODULAR CHASSIS provides 40 LT ports in hardware slots and provides 64 ports in software port allocation (40LT ports and 24 virtual ports). There are 2 types of MODULAR CHASSIS; "Physical MODULAR CHASSIS" and "Virtual MODULAR CHASSIS ". The Physical MODULAR CHASSIS is a “hardware MODULAR CHASSIS” and is used to accommodate an MP, FPs, IP PADs, legacy LT/AP cards, and power supply units. The Virtual MODULAR CHASSIS is “software MODULAR CHASSIS” and is used to accommodate IP stations by system data programming. The port capacity of the Virtual MODULAR CHASSIS is varied depending on the number of Physical MODULAR CHASSIS. The MODULAR CHASSIS can be installed on the desktop or into the 19-inch rack only.

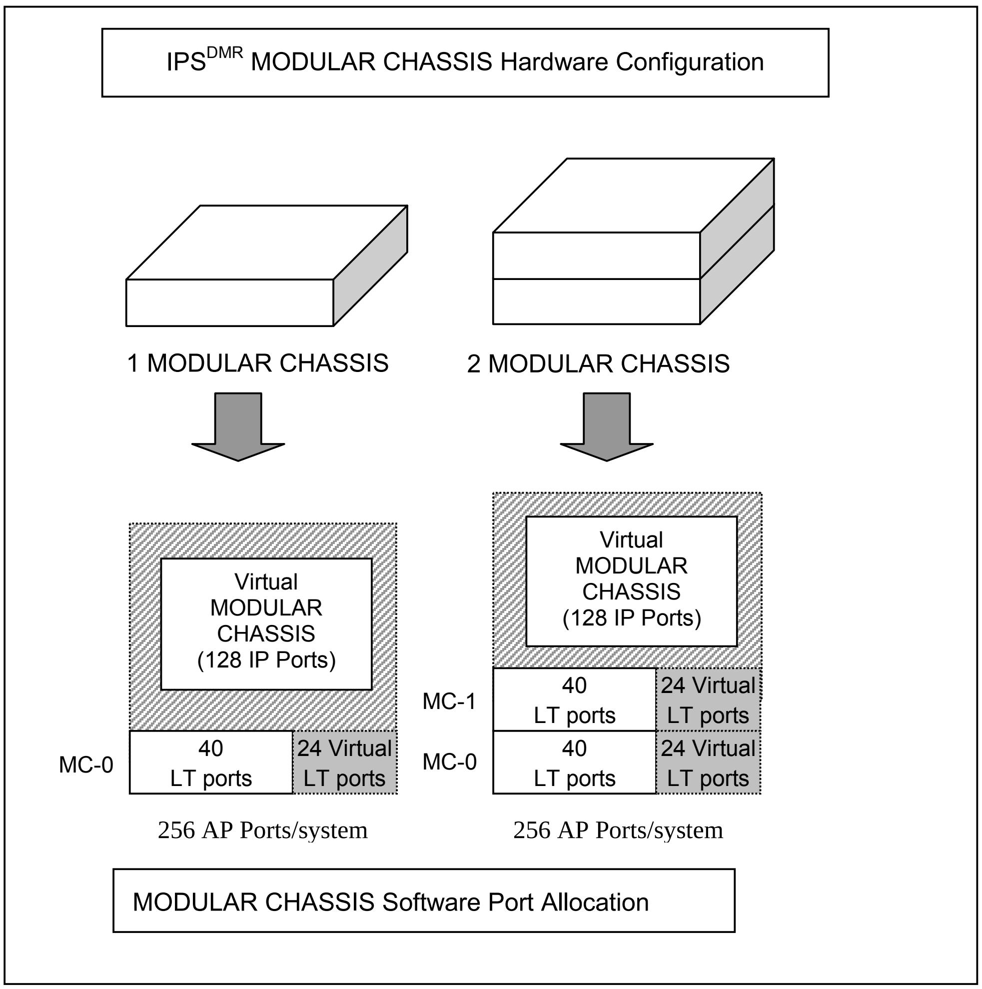
One MODULAR CHASSIS provides 6 card slots including one card slot for Main Processor (MP)/Firmware Processor (FP) and other 5 slots for Line Trunk (LT)/Application Processor (AP) cards; 40LT ports and 24 virtual LT ports; AC, LTC, BUS cable connectors and power switch which are located at the rear side of MODULAR CHASSIS. The following illustration shows MODULAR CHASSIS hardware configurations, software port allocation, face layout and rear view of MODULAR CHASSIS for IPS^{DM}.



NEAX IPS^{DMR} Modular Chassis (MC)

There are 2 types of MODULAR CHASSIS; "Physical MODULAR CHASSIS" and "Virtual MODULAR CHASSIS ". The Physical MODULAR CHASSIS is “hardware MODULAR CHASSIS” and is used to accommodate an MP, FPs, IP PADs, legacy LT/AP cards, and power supply units. The NEAX IPS^{DMR} can consist of one or two MODULAR CHASSIS depending on the system configuration. The Physical MODULAR CHASSIS provides 40 LT ports in hardware slots and provides 64 ports in software port allocation (40LT ports and 24 virtual ports). The Virtual MODULAR CHASSIS is a “software MODULAR CHASSIS” with a port capacity of 64 ports. A maximum of two Virtual MODULAR CHASSIS can be assigned per remote site for a total of 128 ports used to accommodate IP stations by system data programming. The maximum number of Remote Sites is 15. This system locates the maximum of 64 FP/AP cards per system, at multiple Remote Sites. The number of FP/AP cards accommodated at one Remote Site should be a maximum of eight including the MP built-in FP. If more than eight FP/APs are assigned, the system does not operate normally. The MODULAR CHASSIS can be installed on the desktop or into the 19-inch rack only.

The following illustration shows MODULAR CHASSIS hardware configurations, software port allocation, face layout and rear view of MODULAR CHASSIS for IPS^{DMR}.



Remote PIM over IP System Outline

When IPSDMR and 2000 IPS PIM are installed at Remote Site, and connected to a Main Site over IP network, the Main Site system controls and maintains the operation of Remote Sites as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

IPSDMR: IPS Distributed Model Remote (with PN-CP31-A/PN-CP31-B)

IPS PIM: IPS (with PN-CP24-A/PN-CP24-B)

Outline of System Operation

- The MP card of Main Site controls call processing, and Remote Site follows the Main Site in normal operation mode.
- Remote Site can accommodate analog Single Line telephones, Dterm, PS, DtermIP, and LT/AP cards.
- Local Switch (TDSW) at Remote Site connects an outside party when the Remote Site is directly connected to the PSTN/GSTN.
- In the case of connections between Main Site and Remote Site, or Remote Site and Remote Site, the voice path is connected via Peer-to-Peer or IP-PAD.
- If the communications between Main Site and Remote Site are interrupted, the Remote Site starts a survival mode operation after the system reset.

Outline of Survival Mode Operation

- Remote Site system watches a Keep Alive signal sent from Main Site regularly.
- If a line failure occurs (Keep Alive signal is not received), Remote Site resets the own system and starts survival mode operation as a stand-alone system to control the call processing within the Remote Site.
- During survival mode operation, Remote Site system checks regularly whether the communications with Main Site is possible or not. When the Remote Site regards that the communications are possible, the Remote Site will change over to the normal mode to communicate with the Main Site automatically or manually.

Outline of Survival Mode Operation

- This feature displays the link state between Main Site and Remote Site on the designated Dterm or DtermIP at both sites, and allows users at both sites to notice the link failure.

Remote PIM over IP System Conditions

General Conditions

- We recommend that you should install the same version of software in the MP cards of Main Site and Remote Sites.
- The way of loading and conditions of Key FD are following.
  - The same number of Key FD for remote site license as the number of remote sites is required for Remote PIM over IP system.
  - The required Key FD for the whole system must be loaded to the Main Site. If the Key FD is loaded to the Remote Site, the Key FD data is invalid.
  - Total number of terminals that can be accommodated in the Remote Site depends on the number of port and number of license allocated from total Key FD data of Main Site. The total Key FD data is loaded to Main Site and divided between Main Site and Remote Sites as following examples.

    Example 1: When 256-port Key FD is loaded to the Main Site and 64 ports are used in the Remote Site. Remote Site: 64 ports are allocated from the Main Site. Main Site: 192 ports are available

    Example 2: When 128-IP license Key FD is loaded to the Main Site

    The license is allocated in order of connecting IP telephone regardless of Main Site or Remote Site.
  - When the number of port and number of license of Key FD data are insufficient, it causes the FP operation failure or the IP telephone connection failure.
  - After Key FD data is loaded to the Main Site, the service feature, number of port, and number of license to be used by each Remote Site are sent to the flash memory of each Remote Site's MP card automatically. When Remote Sites operate with survival mode, this data is used.
  - Data to be used by each Remote Site is stored in the flash memory every 10 minutes. When a Remote Site starts operating with survival mode within 10 minutes from starting up, the system does not operate normally because the data is not stored in flash memory. The system should be observed for more than 10 minutes after starting up.
  - The number of legacy terminal that depends on the number of PIM in TDSW system depends on the number of port in Remote PIM over IP system. For Key FD data, one PIM license for TDSW system is converted to 64 ports for Remote PIM over IP system.
- The number of accommodated terminals/trunks in Main Site and Remote Site should be a maximum of 1020 ports in the whole system.
- The TCP/IP network is required between Main Site and Remote Site. The closed and bandwidth guaranteed network is preferable, such as IP-VPN (Layer 3 VPN) or wide area Ethernet service (Layer 2 VPN). The following table shows the permissible delay time in the network.

Recommended/ Maximum Value	Permissible delay time	
	One-way	Return-way
Recommended	100 ms.	200 ms.
Maximum Value	120 ms.	240 ms.

  If the network is short of the requirement, it may cause the delay operation of system, the delay and deterioration of voice packets, disconnection of calls, and frequent changeover to survival mode at Remote Site.

- The MP card of Remote Site has the same system data as that of Main Site, because Remote Site automatically gets the data from Main Site when Remote Site starts. In normal operation, Main Site automatically copies the system data to Remote Site through the network at the time set by CM43 Y=7 once a day.
- This feature is available in the Retrofit System (used as Main Site). The system that using the following MP cards can be mixed used in Remote PIM over IP system in any combination.

  Main Site: PN-CP24-A/PN-CP24-B, PN-CP27-A, PN-CP26-A, PN-CP28-A

  Remote Site: PN-CP31-B, PN-CP24-A/PN-CP24-B, PN-CP27-A, PN-CP26-A, PNCP28-A

  NOTE: *PN-CP27-A/PN-CP28-A cannot be used as Backup CPU system in Remote Site.*
- This feature is not compatible with Fusion service of 2400 IPX.
- The RUN lamp of Remote Sites' MP card flash at 60 IPM in normal mode and 120 IPM in survival mode.

Conditions for System Configuration

- The number of Remote Sites is a maximum of 15.
- The total number of FP/AP at Main Site and all Remote Sites should not exceed 64 including the MP built-in FP, Virtual FP, and Virtual IPT.
- The number of FP/AP accommodated at one Remote Site should be a maximum of eight including the MP built-in FP.
- Remote Site can accommodate the following FP/AP/LT cards.
  - FP: One MP built-in FP and two Virtual FP
  - AP: BRT, 24DTI, 24PRT (ISDN-PRI)
  - LT: 8IPLA, DLC, LC, COT, ODT

    NOTE: *Attendant Console, Desk Console, Add-On Module, DSS/BLF Console are not mountable at Remote Site.*
- Remote Site cannot accommodate the following FP/AP/LT cards.
  - FP: FP card (CP15/CP19) is not mountable at Remote Site.
  - AP: AP00 (SMDR), AP00 (DBM), ICH, CCH, DCH (ISDN/Q-SIG/Q931a), 4RSTB (MFC/T1-ANI/E911), 4RSTC/D (Caller ID trunk), CS00 (ATI), CFTC (32-party conference), IPT
  - LT: 8RSTA/G (PBR), ILC (ISDN Terminal), 4RSTE/F (Caller ID station), CFTA/B (6/10-party conference), AMP, 4VCT, DK00, DAT

    NOTE: *Four-line built-in PBR on the MP card is available at Remote Site.*
- When Remote Site is IPSDMR, in addition to the above conditions, the following LT cards are restricted.
  - PN-4LDTA (LDT), PN-4LLCB (LLC), PN-8PFTB (PFT)
- MP built-in FP must be set to each Remote Site.
- 128 ports are assigned per FP card/MP built-in FP in initial, but the port allocation for FP card/MP built-in FP can be changed by the programming (CM05 Y=3). The port for FP card/MP built-in FP must be assigned with every 8 port. If the assignment value cannot be divided by 8, the value that the remainder is omitted is assigned to the FP card/MP built-in FP automatically.

- The available IP-PAD card in the Remote Site is only PN-32IPLA-A/PN-8IPLA.
- For the system capacity of ISDN system or CCIS system, refer to the system capacity of each system manual. Set the whole Remote PIM over IP system within the system capacity and be sure to mount the handler card in the Site where the interface card is installed.
- The number of terminals/trunks accommodated at one Remote Site should be a maximum of 256. This is the sum of maximum 128 physical ports (one built-in FP), maximum 128 IP ports (two Virtual FP), and AP ports such as ISDN, etc.
- Following is connected to the LAN port of the MP card at Main Site.

  Peer-to-Peer CCIS, OAI, MAT
- The highway channel for LT card is allocated to the Main Site with 512 ports and each Remote Site with 128 ports.

Conditions for Survival Mode at Remote Site

- Remote Site starts survival mode operation in the following cases.
  - When the communications (Keep Alive signal) in every 30 seconds between Main Site and Remote Site are interrupted for the predetermined time set by CM0B Y=31-60>50 on normal mode operation.
  - When Remote Site cannot be connected to Main Site or is not allowed connecting to Main Site after the system reset of the Remote Site.
- Remote Site is reset automatically to change the operation from normal mode to survival mode when it detects an interruption of the communications from/to Main Site.
- When Remote Site starts the survival mode operation, the fault information “Initialize by CAT or MAT” (Fault occurrence kind No. 01) is registered to the MP card of Remote Site. In addition, “Communication error occurrence between Main Site and Remote Site” (Fault occurrence kind No. 42) is registered to the MP card of Main Site at 20 seconds later from the predetermined time set by CM0B Y=31-60>50.
- Remote Site on survival mode checks at every 30 seconds if the communications to Main Site are possible. When the Remote Site regards that the communications are possible, “Communication error restoration between Main Site and Remote Site” (Fault occurrence kind No. 52) is registered to the MP card of Main Site at 20 seconds later from the predetermined time set by CM0B Y=31-60>51.
- When Remote Site on survival mode regards that the communications to Main Site are possible, manual changeover (system reset) is required. Automatic changeover (re-connection to Main Site) is also selectable in system data programming (CM0B Y=31-60>51, 53). At the automatic changeover, Remote Site system is initialized and the calls on going are disconnected due to the reset of terminals.
- During survival mode operation, it is not possible to originate/terminate calls to/from stations at other sites.

Service Conditions

- Set the unique location number to each location group for proper setting of the IP-PAD channel selection and CODEC for voice compression.
- The system clock at Remote Site synchronizes with the system clock at Main Site. If the communications to Main Site are interrupted, Remote Site does not synchronize and operates at the hardware clock on the MP card of Remote Site.
- Remote Site cannot accommodate the Virtual IPT and the IPT card (H.323).
- A different metering area at each Remote Site is not available because Main Site corrects all call metering through the whole system. If a call is originated from the COT at Remote Site, it is charged as the call originated from the COT at Main Site.
- The service features requiring continuous voice transmission such as Background Music and Internal Zone Paging should not be used at Remote Site considering the traffic on the network.
- Multiple line service should not be used among different sites considering the traffic on the network.
- Automated Attendant should not be used at Remote Site considering the traffic.
- Unavailable services at Remote Site are as follows.
  - Attendant Console, Desk Console, Add-On Module, DSS Console and ISDN Terminal are not mountable.
  - Caller ID - Station is not available.
  - CCIS (CCH with digital and analog trunk interface) is not available. CCIS (CCT) is available.
  - All services using MP built-in DK or DK card are not available.
  - All services using DAT card are not available.
- When Remote Site is IPSDMR, in addition to the above conditions, the following services are restricted due to the PN-CP31-A/PN-CP31-B hardware conditions.
  - MP built-in Modem communications are not available.
  - RS Port No.1 is not available.
  - MP built-in Digital Announcement Trunk (DAT) service is not available.
- Station number of the stations accommodated in Remote Sites should be divided with the Tenant (CM23) according to the PSTN/GSTN or private line to be connected, or each service feature to be used.
- Set the unique trunk route to each Site.
- Different numbering plan for every site must not be assigned.
- Since Main Site controls OAI, OAI terminals cannot be used at Remote Site during survival mode operation.
- Stations at Remote Site cannot participate in the 6/10-party conference using CFTA/CFTB card.
- Each Remote Site must provide a hold tone to the station at the Remote Site.
  - External hold tone (using Jack on the MP card): Tone source is required for each site.
  - Internal hold tone: Available
  - Hold tone using DAT: Not available
- The maximum station number which can be used in SMDR or PMS Interface is as follows.
  - MP built-in SMDR/PMS on IP: 1020 stations
  - SMDR/PMS with AP00 on RS-232C: 504 stations

Maintenance Conditions

- You can connect MAT to Remote Site via RS-232C or LAN.
- Do not set and change the system data at Remote Site, except setting the Remote Site number by CM0B Y=00>90. If the system data is set and changed at Remote Site, normal operation is not guaranteed. In addition, the system data copy from Main Site overwrites the system data in Remote Site once a day.
- SNMP is not available at Remote Site.
- MP program download is available via RS-232C at each site.
- Resident system programming must not be used.
- The IPSDMR at Remote Site does not have DHCP server and client function.
- AP program download is not available for the AP cards at Remote Site.
- The MJ/MN alarm indications are not available at Remote Site. If a fault occurs at Remote Site, the fault is notified to the Main Site and MJ/MN alarm is indicated at the Main Site.
- On-line expansion for LC/DLC/COT cards is supported and it enables the tone/path connection even if the system data copy is not activated to the Remote Site. After completing all expansions for LC/DLC/COT cards, be sure to execute the system data copy to the Remote Site (CMEC Y=8). Note that the expanded data will not be added if the Remote Site starts survival mode operation before the system data copy.
- After changing the IP address (CM0A Y=01) and TCP base port number (CM0A Y=10-17/30-37/100-115) of IP-PAD, the changed data is reflected when the Make Busy key of the corresponding IPPAD card is set to ON then OFF, even if the system data copy is not executed. This is only available for change of IP address and TCP base port number of IP-PAD.

Conditions for Link Down Notice

- Link Down Notice is available only for Dterm and DtermIP accommodated in 2000 IPS and IPSDM/IPSDMR. This is not available for a single line telephone and Attendant Console.
- For message display, Dterm/DtermIP with 24-digit or more LCD is recommended. 16-digit LCD may not display all messages properly.
- Notification message can be displayed regardless of idle or busy state of Dterm/DtermIP, writing the message over the present display. After six seconds, the display returns to the time display automatically.
- The system detects a Link Down on the condition that UDP connection between Main Site and Remote Site is interrupted. The Link Down is notified to the Dterm/DtermIP at 20 seconds later from the time set by CM0B Y=31-60>52. If the time is not set by CM0B Y=31-60>52, the Link Down is notified to the Dterm/DtermIP at 20-50 seconds later from the system detects the Link Down.
- When the link between Main Site and Remote Site is interrupted, the lamp of Dterm/DtermIP button becomes the state as shown below. Then press the button, the LCD of the Dterm/DtermIP displays the following.

COLOR AND STATE OF BUTTON		STATE AND OPERATION	LCD DISPLAY
Red/Flashing (Momentarily)	0.125 seconds ON- 0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON- 0.5 seconds OFF	Press the button after Link Down occurrence	Link Down to Site xx (xx: Site No.)
Green/Flashing (Intermittently)	0.125 seconds ON- 0.125 seconds OFF- 0.125 seconds ON- 0.625 seconds OFF	Remote Site is survival mode after Link restoration	-
Green/Flashing (Intermittently)	0.125 seconds ON- 0.125 seconds OFF- 0.125 seconds ON- 0.625 seconds OFF	Press the button with Remote Site is survival mode after Link restoration	Link Down to Site xx (xx: Site No.)
OFF	-	Remote PIM is normal mode after Link restoration	-
OFF	-	Press the button with Remote Site is normal mode after Link restoration	Normal Condition: R-PIM

- Link restoration is notified to the Dterm/DtermIP at 20 seconds later from the time set by CM0B Y=31-60>51. If the time is not set by CM0B Y=31-60>51, the Link restoration is notified to the Dterm/DtermIP at 110-150 seconds later from the system detects the Link Ready. After the link is ready, the lamp of Dterm/DtermIP button keeps flashing during the Remote Site operates on survival mode. Since the color of lamp and the indication interval changes, an administrator at the Remote Office can changeover the system operation from survival mode to normal remote mode according to this indication. When the link between Main Site and Remote Site recovers and the Remote Site starts subordinate operation to the Main Site (normal mode), the flashing lamp of the button goes out.

Chapter 3 Terminals

A variety of terminal equipment may be connected to the NEAX 2000 IPS. The following equipment may be installed with the system.

- SN716 DESK CON (Attendant Console)

- Dterm Series i Analog Terminals
  - Single-Line Analog
  - Hospitality Single-Line Analog

- Dterm Series i (IP) Terminals
  - 4-Line display Dterm IP
  - 8-Line display Dterm IP
  - 16-Line display Dterm IP
  - 16LD display Dterm IP
  - 32-Line display Dterm IP

- Dterm Series i (TDM) Digital Terminals
  - 2-Line Digital
  - 4-Line display Digital
  - 8-Line non-display Digital
  - 8-Line display Digital
  - 16-Line display Digital
  - 16LD display Digital
  - 32-Line display Digital
  - 60 Console Add-On Module/DSS/BLF

- Dterm Series E (TDM) Digital Terminals
  - 2-Line Digital
  - 8-Line non-display Digital
  - 8-Line display Digital
  - 16-Line display Digital
  - 32-Line display Digital
  - 60 Console Add-On Module/DSS/BLF

- Dterm Cordless Terminals
  - Dterm Handset
  - Dterm Cordless Lite
  - Dterm Cordless II
  - Dterm Analog Cordless

- PS - Wireless Handset

- Dterm SP20 (IP Soft Phone)
- Dterm SP30 (IP Soft Phone)

- Inaset

SN716 Desk Con

The SN716 Desk Console has an ergonomic design and provides full access to all PBX Console features. It connects to the NEAX 2000 IPS using the same circuit cards as the Dterm Series i/E terminals. The SN716 Desk Console operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and reentering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console. The SN716 Desk Console also provides flexible key assignments to meet the operator's needs. Key assignments are semi-fixed by system default data, but may be changed via programming.

The NEAX2000 IPS supports a maximum of eight SN716 DESK Consoles.

SN716 DESKCON Features

- Character LCD (4x40 characters)
- LCD designation strips
- Software-controlled LCD loop key
- Full access to PBX features
- Headset connectivity
- Recorder connectivity

Dimensions: 10 inches (25.4 cm wide x 9 inches (22.9 cm) deep x 4 inches (10.2 cm) high

Line Conditions of the SN716 DESKCON

The cable length between the DLC card and terminal varies depending on the type of terminal. The table shows the line conditions of the Attendant Console.

Line Conditions of the SN716 DESKCON

Interface Cards Type	Power Options	Cable Length* (Cable 0.5/24 AWG)
PN-8DLCL/8DLCP	PN-PW00	1000 ft. (304 m)
	AC Adapter	
PN-4DLCM/4DLCQ	PN-PW00	1000 ft. (304 m)
	AC Adapter	4000 ft. (1200 m)

*Cable length is based on the diameter of the cable and the terminal impedance.

SN716 DESKCON Exclusive Features

While the DESKSON has full access to PBX features the SN716 DESKCON has the following exclusive features.

DESKCON Exclusive Features	
Attendant Assisted Calling	Call Waiting Display
Attendant Camp-on (Full and Semi-automatic)	Common Route Indial
Attendant Called/Calling Name Display	Dialed Number Identification Service (DNIS)
Attendant Called/Calling Number	Incoming Call Identification
Attendant Call Selection	Individual Trunk Access
Attendant Console Lockout-Password	Multi-Function Key
Attendant Do Not Disturb Setup and Cancel	Multiple Console Operation
Attendant Interposition Calling/Transfer	Pushbutton Calling - Attendant Only
Attendant Lamp Check	Serial Call
Attendant Listed Directory Number	Time Display
Attendant Loop Release	Trunk Group Busy Display
Attendant Programming	Unsupervised Trunk-to-Trunk Transfer By Attendant
Attendant Training Jacks	Attendant Delay Announcement
Audible Indication Control	Attendant Lockout
Call Processing Indication	Attendant Overflow
Call Queuing	Attendant Override
Call Splitting	

Note: *For Detail of each feature refer to Chapter 9 Feature Description.*

Functions and use of Keys and Lamp Indications

LOCATION NUMBER	KEY OR LAMP DESIGNATION	FULL NAME	BASIC /OPTION	KEY /LAMP	FUNCTION
1	L1-L6	Loop	Basic	Key	The attendant answers the call associated to the particular loops. Loop keys are usually used to reenter to held calls, answer automatic recalls.
2	L1A-L6A	Loop Lamp A	Basic	Lamp (one per loop)	Steady green lamp indicates attendant connected to the loop, or called station has answered. Flashing green lamp indicates call waiting to be answered. Steady red lamp indicates call party busy. Flashing red lamp indicates call held at the console.
3	L1B-L6B	Loop Lamp B	Basic	Lamp (one per loop)	Flashing red lamp indicates automatic recall has been activated.
4	Push-button Dial	Push-button Dial	Basic	Key	Allows the attendant to: • Process incoming calls • Originate calls • Activate various service features
5	SRC	Source	Basic	Key & Lamp	Allows the attendant to speak with the calling party. The associated lamp lights when the attendant is connected. The source trunk/station number will be shown in the number display field.
6	DEST	Destination	Basic	Key & Lamp	Allows the attendant to speak with the called party. The associated lamp lights when the attendant is connected. The destination station/trunk number will be shown in the number display field.
7	Talk	Talk	Basic	Key	Allows the attendant to join in a three-way conference with the calling and called parties. When connection is established, both SRC and DEST lamps will light.
8	Cancel	Cancel	Basic	Key	Allows the attendant to: • Disconnect the calling (source) or called party (destination) from the loop. • Disconnect an outgoing trunk or tone seized by the attendant. • Disconnect the station recalling attendant for transfer assistance.
9	Hold	Hold	Basic	Key	Allows the attendant to hold a call at the console and/or to activate it to serial call state.
10	Release	Release	Basic	Key	Allows the attendant to release from an established connection freeing the console for processing of new calls.
11	Answer	Answer	Basic	Key & Lamp	Allows the attendant to answer incoming calls in the order in which they arrive at the console.
12	Start	Start	Basic	Key & Lamp	Allows the attendant to extend an outgoing call to a station. Completion of outgoing pulse will be recognized.

Functions and use of Keys and Lamp Indications (Cont.)

13	Incoming Call Identification: These eight non-locking keys with associated lamps provide attendant access to specific types of incoming calls. A flashing lamp indicates a call waiting to be answered. A steady lamp indicates a call answered. The standard arrangement of these keys is shown on the face layout.				
	LDN	Listed Directory Number	Basic	Key & Lamp	Incoming central office trunk call.
	TIE	Tie Line	Basic	Key & Lamp	Allows the attendant to answer incoming Tie Line calls when the distant station dials access digit to the attendant.
	Busy	Call Forwarding-Busy Line	Basic	Key & Lamp	Allows the attendant to answer incoming calls to specified station when the station is busy.
	ATND	Attendant	Basic	Key & Lamp	Incoming station call.
	NANS	Call Forwarding-Don’t Answer	Basic	Key & Lamp	Allows the attendant to answer incoming calls to specified station when the station does not answer within the predetermined time.
	Recall	Recall	Basic	Key & Lamp	Incoming station call for attendant assistance in transferring an established outside call to another station.
	Option	Optional	Option	Key & Lamp	Additional incoming special service calls, such as FX.
	Option	Optional	Option	Key & Lamp	Additional incoming special service calls, such as FX.
14	PAGE	Page	Basic	Key & Lamp	Allows the attendant to connect with pager. (overhead paging)
15	REC	Record	Basic	Key & Lamp	Allows the attendant to connect with recorder.
16	EMG	Emergency	Basic	Key & Lamp	Allows the attendant to answer incoming calls from the station where the station leaves the receiver off.
17	BV	Busy Verification	Basic	Key & Lamp	Allows the attendant to enter into station-to-station connection.
18	TRKSL	Trunk Selection	Basic	Key & Lamp	Allows the attendant to individually select a desired trunk.
19	Call Park	Call Park	Basic	Key & Lamp	Allows the attendant to establish incoming call to Call Park. **Note:** *The attendant can connect this call once again by dialing a specific number and individual number of the console from an ordinary extension telephone.*
20	SC	Serial call Set	Basic	Key & Lamp	Allows the attendant to be automatically recalled when the station user replaces the handset, by depressing the key after extending a central office incoming call to the station user.
21	SVC	Supervisory Call Set	Basic	Key & Lamp	Allows the attendant to supervise a call by depressing the key after extending a central office incoming call to the station user.
22	Option	Optional	Option	Key & Lamp	Allows the attendant access to various optional features provided as required. Each key has an associated lamp.
23	Mute	Mute	Basic	Key & Lamp	Allows the attendant to cut off the voice transmission.
24	Alarm	Alarm	Basic	Key	Steady lamp indicates trouble conditions in the PBX.
25	Position Available	Position Available	Basic	Lamp	When the attendant position (console) is available to process calls, the lamp lights.

Functions and use of Keys and Lamp Indications (Cont.)

26	Position Busy	Position Busy	Basic	Key & Lamp	When the key is pressed, the lamp will light, and the console will become not available. **Note:** *Press the button if operators leave their seats.*
27	Night	Night	Basic	Key & Lamp	Allows the console to place in the night answer mode and lights the associated lamp. Releasing the key restores the console to normal operation.
28	Up Down	Up and Down	Basic	Key	Allows the attendant to adjust volume of the receiver, ringer and contrast of the LCD.
29	LCD	Liquid Crystal Display	Basic	Display	The following information will be displayed: 1st line: The kind of party that connects to the attendant, the number of the waiting calls, the date and time. 2nd line: Tenant number, station class of service and station number belonging to the destination (called) party. 3rd Line: Tenant number, station class of service and station number belonging to the source (calling) party. 4th Line: Optional indication, such as trunk busy.
30	Multi-Function Key	Multi-Function Key	Basic	Display	Keys while idle: Mode—allows access to DAY/NIGHT mode and LKOUT (Console Lock out mode) Prog—allows access for programming DISA, System Speed Dial, Date & Time and Tone Ringer Keys while answering or originating: SPB—Out Pulse Short LPB—Out Pulse Long SHF—Flash Over Trunk Keys while calling a busy station: B.V—Busy Verification Keys while calling a DND station: DDOVR—DND override Keys while accessing Hotel feature: RC—Room Cut off MW—Message Waiting DD—Do not Disturb WU—Wake up call RESET—Reset

Dterm Series i Analog Terminals

The Dterm Series i is available in two distinct analog models.

Single-Line

Hospitality Single-Line

Descriptions of the Analog Terminals

DTR-1 (WH) TEL	Fully modular with Redial key, Flash key, Message Waiting lamp, Data Jack and Ring/Handset Receive Volume.
DTR-1 (BK) TEL	
DTR-1HM (WH) TEL	Fully modular with Redial key, 'Flash' key, Message Waiting Lamp, Data Jack, eight programmable Feature/Speed Dial keys and Ring/Handset Receive Volume.
DTR-1HM (BK) TEL	

Line Conditions of the Analog Terminals

Terminal Type	Card Type	Cable Length* (Cable 0.5/24 AWG)
DTR-1 DTR-1HM	PN-4LCD-A (max. 600 ohms loop resistance)	Approximately: 1.43km (.88 miles)
	PN-8LCAA (max. 600 ohms loop resistance)	Approximately: 1.43km (.88 miles)
	PN-AUCA (max. 2500 ohms loop resistance)	Approximately 12.29km (7.63 miles)
	PN-4LLCB (max. 2500 ohms loop resistance)	Approximately 12.29km (7.63 miles)

*Cable length is based on the diameter of the cable and the terminal impedance.

Specification for Analog Terminals

Item	Description
Size	224mm x 165mm +/5mm (Length x Width)
Color	Black or White
Dial Pad	12-Key Dial Pad: 4 Rows and 3 Columns; Metropolitan Dial Pad with Alphabet, * and # buttons; Button 5 has a Raised Dot
Type of Dial	DTMF and Dial Pulse
Function Buttons	Hook flash, Redial Key on DTR-1-1; Hookflash, Program, Redial, Monitor, and Hold Keys on DTR-1HM-1
Message Waiting Lamp	Neon Lamp with Window Design -Glow Through Filter Raised from Surface with MW and Incoming Ring Indication
Operating Voltage	Activation Voltage 88V to 108V, Deactivation Voltage 53V or Less
Speed Dials	(DTR-1HM Only) 8 Buttons, Maximum 21 Digits
Hookflash Timer	630+/10ms (Fixed)
Redial Key	Maximum of 31 digits
Ring Vol. Control	3 Levels (Soft, Medium, Loud) Programmable
Ring Tone Pitch Control	3 Levels (Slow, Medium, Fast, Off) Programmable
Handset Receiver & Speaker Volume	6 Levels (Volume Key)
Handset	Hearing Aid Compatible, Dynamic Type Element
Handset Cord	12 feet
Directory Card	Large Convenient Directory Card
Data Jack	Dedicated Jack; Used for Connection to Modem, Speakerphone, etc., located on back of telephone
Wall-Mount Unit	Built-in
Electrostatic Discharge	Can Withstand +/- 20kv Discharge
Approvals	c-UL (Ul 60950 3rd Edition) FCC part 15, 68, IC (Industry Canada)

Dterm Series i (IP) Terminals

The Dterm IP gives you the freedom to tailor your platform and telephony applications for your business even as the business grows and your needs expand. With its advanced digital circuitry, the Dterm IP consists of several distinct models to meet users' diverse telephone terminal needs.

Dterm IP terminals are designed to provide ergonomic form and user-friendly functions. Dterm IP terminals offer an adjustable LCD display unit with menu-driven soft key operation, allowing users to program terminals at the desktop. The LCD panels are equipped with three lines of display, each with 245 characters. Standard features include headset jacks, wall mounts and adjustable base units.

Dterm IPs has four soft keys located just under the display of each terminal. These menu-driven soft keys allow users convenient access too many telephony features. According to the status of the multi-line terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the terminal changes, the soft keys displayed will change automatically.

Dedicated function keys provide easy one-touch access to the most common telephone operations. These keys include: feature, recall conference, redial, hold, transfer, answer, speaker, microphone, directory and message.

Dterm IP terminals are ideal choices for users that are connected through a managed IP network. Dterm IP terminals are class B devices and comply with U.S. FCC regulations for office and residential use as well as the Canadian Inference-causing Equipment Regulations.

8-Button Display

16-Button Display

32-Button Display

16-Button LD

4-Button Display

Dterm IP Terminal Features

Dterm IP 4D

- 4 Line/Feature Access/ Programmable Feature Access Keys
- 9 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

Dterm IP 8D

- 8 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

Dterm IP 16D

- 16 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

Dterm IP 16LD

- 16 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

Dterm IP 32D

- 32 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

Dterm IP Terminal Features

Display Features	
Liquid Crystal Display (LCD)	All Models: 3 lines by 24 Characters
Brightness Control LCD Contrast	All Models: Yes
Adjustable LCD Display	Dterm 8, 16, 16LD, 32 models only
Backlit Display Compatible	Dterm 8, 16, 16LD, 32 models only
Receiver Volume Control	
Handset	All Models: Yes
Full Duplex Speaker Phone	All Models: Yes
Ring Volume Control	All Models: Yes
Miscellaneous	
On-Line Firmware Upgradeable	All Models: Yes
DHCP	All Models: Client Support
Call Message Indicator	All Models: Yes
Headset	Dterm 8, 16, 16LD, 32 models only
Supported Adapters	AD(A)-2R (Local Recording) PS(A)-R (Local Line Survivable)
Adjustable Base	All Models: Yes* **For Dterm 4D - Terminal height can be adjusted with removal/addition of the base unit.*
Built-in Wall Mount	All Models: Yes
Built-in Headset Jack	Dterm 8, 16, 16LD, 32 models only
Housing Color: Black	All Models: Yes
Housing Color: White	Dterm 8, 16, 16LD, 32 models only

Dterm IP Terminal Specifications

Network Parameters	
Internet Layer	All Models: IPv4
IP Protocol	All Models: NEC Peer-to-Peer (only)
Jitter Buffer	All Models: Max 300msec (10msec steps)
Payload Interval	All Models: 10ms ~ 40ms (10ms steps)
IP Addressing	All Models: DHCP or Static Assignable
QoS	All Models: 801.p, ToS and Diff-Service
Power Support Options	
External	All Models: AC: 24V, Current: 750mA
Operating Temperature	All Models: 0 - 40 deg C (32 - 103 deg F)
Spare Pair Power	All Models: Yes
In-Line Power	All Models: Yes
External Power via AC adapter	All Models: Yes (Optional adapter)
Quality of Service	All Models: Yes Layer 2: 802.1p/Q; Layer 3: IP Precedence, Diff-Services

Equipment Specification Size (W x D x H)	
Dterm IP 4D	Tilt up: 7.57" x 8.69" x 5.57"
	Tilt down: 7.57" x 8.69" x 3.80" (without stand unit)
	Weight: 1.98lbs
Dterm IP 8D	Tilt up: 9.09" x 8.54" x 5.28"
	Tilt down: 9.09" x 8.54" x 4.17"
	Weight: 2.51lbs
Dterm 16D	Tilt up: 9.09" x 8.54" x 5.28"
	Tilt down: 9.09" x 8.54" x 4.17"
	Weight: 2.51lbs
Dterm IP 16LD	Tilt up: 9.09" x 9.88" x 5.28"
	Tilt down: 9.09" x 9.88" x 4.17"
	Weight: 2.91lbs
Dterm IP 32D	Tilt up: 9.09" x 9.57" x 5.28"
	Tilt down: 9.09" x 9.57" x 4.17"
	Weight: 2.84lbs

Dterm IP Terminal Specifications

Dterm IP 4D
- Voltage: 48V
- Current: 90ma
- Power Consumption: 4.32W
- Audio Algorithm: G.711, G.729A
- Protocol Support: 802.3af (CDP and NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45

Dterm IP 8D
- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

Dterm IP 16D
- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

Dterm IP 32D
- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

Dterm IP 16LD
- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

Dterm Series i (TDM) Digital Terminals

The Dterm Series i Terminals are designed to provide ergonomic form and user-friendly functions. With advanced digital circuitry, the Dterm Series i consists of distinct models to meet users' diverse telephone terminal needs.

Dterm Series i Terminals offer adjustable display and non-display units with menu-driven soft key operation, allowing users to program terminals at the desktop. Standard features include headset jacks, wall mount units and adjustable base units. The display units are equipped with large LCD panels with three lines of display, each with 24 characters. A 16-button backlit display version is available for installations in dimly lighted areas such as restaurants, night clubs, and residential applications. Easy to see in either dark or bright applications, the backlight feature may expand installation opportunities and markets.

The Dterm Series i Display Terminals have four soft keys located just under the display of each Terminal. These menu-driven soft keys allow users' convenient access too many features. The state of the terminal will determine what soft key is available to the user. According to the status of the Multiline Terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the Multiline Terminal changes, the soft keys displayed will change automatically.

Dedicated function keys provide easy one-touch access to the most common telephone operations. These keys include: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, and Message.

All Dterm Series i telephones are an ideal choice for both businesses and remote users in residential home offices. All Dterm Series i telephones are Class B devices and comply with U.S. FCC regulations for office and residential use, and with requirements of the Canadian Interference-Causing Equipment Regulations.

2-Button Non-Display

8-Button Non-Display

8-Button Display

16-Button Display

32-Button Display

60-Line DSS/BLF Console

4-Button Display

16-Button LD

Standard features of the Digital models

- Large Message Waiting LED
- 24 Character, 3 Line LCD on display equipped models
- Tilt LCD Unit
- Adjustable Base
- Built-in Wall Mount Unit
- Built-in Headset Jack Connector
- 6 Programmable Ring Tones
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2 Color LED
- Backlit Display on 16-Button Model
- Four Local Soft key Controls (detail functions are dependent on PBX, only provided on terminals with display)
- Eleven dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Mic*, Directory*, and Message*. (*Functionality dependant upon system software.)
- Built-in Half Duplex Handsfree Unit
- Snap-in Options Available:
  - AP(R): Analog TEL connection with Ringing Signal Generation
  - AP(A): Analog TEL connection without Ringing Signal Generation or Disconnect Signal
  - AD(A): Tape-recorder connection
  - CT(A): CTI Adapter, RS-232-C (9-pin) interface
  - IP-R: VoIP Adapter

Series i Multi-line Terminal Descriptions

<table>
<tr><td>DTR-2DT (BK) TEL</td><td rowspan="2">2 LINE TERMINAL - available in black (BK) or white (WH).
(does not support optional adapters)
Fully modular with 2 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, electronic volume and tone controls.</td></tr>
<tr><td>DTR-2DT (WH) TEL</td></tr>
<tr><td>DTR-4D (BK) TEL</td><td>4 LINE TERMINAL – available in black (BK) only.
Fully modular with 4 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls.</td></tr>
<tr><td>DTR-8 (BK) TEL</td><td rowspan="2">8 LINE TERMINAL – available in black (BK) or white (WH).
Fully modular with 8 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, headset jack, wall mount unit, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand.</td></tr>
<tr><td>DTR-8 (WH)TEL</td></tr>
<tr><td>DTR-8D (BK) TEL</td><td rowspan="2">8 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).
Fully modular with 8 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls, and tilt stand.</td></tr>
<tr><td>DTR-8D (WH) TEL</td></tr>
<tr><td>DTR-16D (BK) TEL</td><td rowspan="2">16 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).
Fully modular with 16 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand.</td></tr>
<tr><td>DTR-16D (WH) TEL</td></tr>
<tr><td>DTR-32D (BK) TEL</td><td rowspan="2">32 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).
Fully modular with 32 Flexible, 2-color LED Line keys, eight Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand.</td></tr>
<tr><td>DTR-32D (WH) TEL</td></tr>
<tr><td>DCR-60 (BK) CONSOLE</td><td rowspan="2">ATTENDANT ADD-ON CONSOLE - Requires an AC-R ADP (included).
Fully modular with 48 programmable, 2-color LED keys (for station trunk appearances), 12 Function keys with red LED, and tilt stand.
Terminal available in: black (BK) or white (WH).</td></tr>
<tr><td>DCR-60 (WH) CONSOLE</td></tr>
</table>

Specification for Series i Digital Terminals

Item	Description
Curl Cord Length	12ft
Weight (no handset)	510g (min.)
LCD Display	24 digit x 3 line (alphanumeric and some characters) no back light
Ringing Sound Level	Max. 80dBSPL (in output limit condition) Max. 86dBSPL (in normal condition)
Built in Hands Free	Half duplex
Items Provided with Instrument	Line cord, Directory card
Handset Cradle	K type compatible
LCD angle	14~42.5 deg. (on the desk, no housing tilt) 25~53.5 deg. (on the desk, housing tilt up) -4.4 deg. (wall mounting)
Housing Face Angle	14~25 deg. (on the desk) -4.4 deg. (wall mounting)
Recommended Headset	NEC Headsets
Other	HAC

Dterm Series i Digital Terminal Options

Item	Description
WM-R Unit	Series i Multi-line Terminals with an AP(R)-R, AP (A)-R, CT (A)-R, and/or an IP-R Unit can be wall mounted using the WM-R Unit.
AD(A)-R Unit	Provides Series i Multi-line Terminals ability to interface with recording device.
AP(R)-R Unit	Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. Provides ringing to analog device connected. Requires an AC-R ADP.
AP(A)-R Unit	Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. No ringing is provided.
CT(A)-R Unit	Connects a Series i Multi-line Terminal to a PC providing a complete turnkey package with graphical telephone user interface and call logging. Shipped with Multi-line Phone Kits software. Supports Serial interface.
IP-R Unit	A compact plug-and-play device that installs into the base of a Dterm Series i display terminal. Integrated two-port 10/100baseT Ethernet pass through hub that permits using one port to connect the network interface card (NIC) from the PC to the IP network. The other is plugged directly into a LAN or an IP network device such as a router, DSL modem or cable modem. Requires an AC-R ADP.
AC-R Unit	AC adapter for DSS/BLF Console, AP(A), AP(R), or IP-R Unit

Dterm Series i Line Conditions

Cable Length Note 2			Standard	with AC Adapter
Dterm Series i	Dterm 8 / 8D	8DLC	300m (984ft)	**Note 2**
		4DLC	300m (984ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	Dterm 16/16D	8DLC	200m (656ft)	**Note 2**
		4DLC	200m (656ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	Dterm 32/32D	8DLC	200m (656ft)	**Note 2**
		4DLC	200m (656ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	DSS/BLF Console **Note 3**	8DLC	-	300m (984ft)
		4DLC	-	1200m (3937ft)
		2DLC	-	1200m (3937ft)

Note 1: Cable length is based on the following conditions.

- *Diameter of the cable is 0.5 mm.*
- The Protection arrester is not inserted between the terminal and PBX.

Note 2: *When using 8DLC card, it is not available for long line function, even if it is equipped with AC Adapter*.

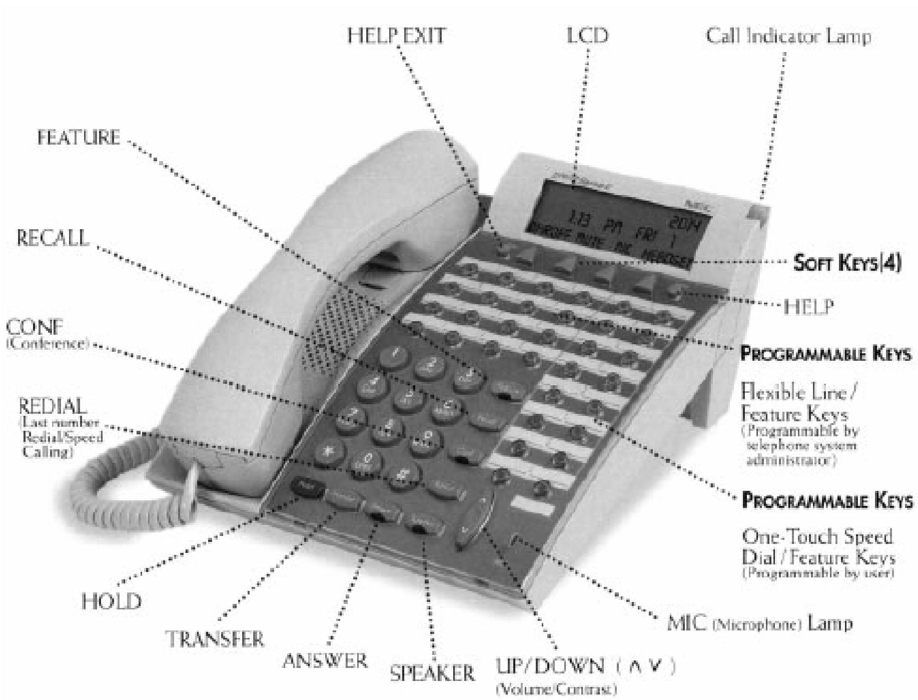
Note 3: The DSS/BLF Console requires local AC/DC supply.

Dterm Series E (TDM) Digital Terminals

The Dterm Series E terminals are designed to provide ergonomic form and user-friendly functions. With advanced digital circuitry, the Dterm Series E consists of distinct models to meet users' diverse telephone terminal needs.

Dterm Series E terminals offer adjustable display and non-display units with menu-driven soft key operation, allowing users to program terminals at the desktop. The display units are equipped with large LCD panels with three lines of display, each with 24 characters. Each terminal offers an optional full duplex speaker phone operation for two-way conversation. Standard features include headset jacks, wall mount units and adjustable base units.

The Dterm Series E Display Terminals have four soft keys located just under the display of each Terminal. These menu-driven soft keys allow users convenient access too many features. The state of the terminal will determine what soft key is available to the user. According to the status of the Multiline Terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the Multiline Terminal changes, the soft keys displayed will change automatically.



Dterm Series E Terminals

8-Line Non-Display

8-Line Display

16-Line Display

32-Line Display

60-Line DSS/BLF/Add-On Module

Dterm Series E Terminals

8-Line Non-Display

- 8 programmable line/feature keys with two-color LED indication
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Message waiting LED
- This terminal is available in Black (BK) or White (WH).

8-Line Display

- 8 Programmable line/feature keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

16-Line Display

- 16 Programmable line/feature keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

32-Line Display

- 16 Programmable line/feature keys with a two-color LED
- 16 SPD/DSS keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

60-Line DSS/BLF/Add-On Module

- 60 programmable line keys (each with a two-color LED)
- This terminal is available in Black (BK) or White (WH).

DTERM SERIES E FEATURES

- Four Local Soft key Controls (detail functions are dependent on PBX)
- Large Message Waiting LED
- 24 Character, 3 Line LCD
- Tilt LCD Unit
- Adjustable Legs
- Built-in Wall Mount Unit
- Built-in Headset Jack Connector
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2 Color LED
- Eight dedicated function keys (Feature, Recall, Conf, Redial, Hold, Transfer, Answer & Speaker)
- Built-in Half Duplex Handsfree Unit
- Optional: Full Duplex HFU (consist of HFU-U Unit, External Microphone Unit and AC adapter)
- Snap-in Options Available:
  - ADA-U (Ancillary Device Adapter)
  - APR-U (Analog Port Adapter with Ringing)
  - HFU-U (External Handsfree Unit)
  - ACA-U (AC Adapter Unit)
  - WMU-U (External Wallmount Unit)
  - WMU-W (External Wallmount Unit for DTP-2 Terminals)

Dterm Series E Digital Terminal Options

Item	Description
ADA-U UNIT	Ancillary Device Adapter, Used for External Recording Devices
APR-U UNIT	Analog Port Adapter with Ringing, used to connect external analog devices such as PC Modem, analog Cordless etc. Requires ACA-U Unit for local power.
APA-U UNIT	Analog Port Adapter without Ringing, Used for external speakerphones. Requires ACA-U Unit for local power when the connected device is over 50 ft. from the APA-U Unit.
HFU-U UNIT	Full duplex Speakerphone with external microphone.
CTA-U	CTI Adapter Unit
ACA-U UNIT	AC Adapter provides local power for APR-U Unit, APA-U Unit, CTA-U Unit, HFU-U Unit, IPW-U Unit, DESKCON, and Dterm long line.
WMU-U UNIT	Wall Mount Unit, Required when using terminal adapters attached to phone. If not using terminal adapters built-in wall mount in phone is used.
IPW-2U (ELC)	The IPW-2U (ELC) adapter is a full duplex switch, which can be installed on any of the Dterm Series E display phones. With this adapter, you can upgrade the 8, 16, 32-button Dterm Series E terminals to VoIP when using the PN-32IPLA (ELC) IP station card.
IPW-2U (P-P)	The IPW-2U (P-P) adapter is a full duplex switch, which can be installed on any of the Dterm Series E display phones. With this adapter, you can upgrade the 8, 16, 32-button Dterm Series E terminals to VoIP when using Peer to Peer connection.

Dterm Series E Line Conditions

Loop Resistance Note 1				
Analog Telephone Set	Standard		600 ohms	
	Option		2,500 ohms	
Loop Start Trunk	Exchange Line		1,700 ohms	
	Tie or DID Line		2,500 ohms	
Cable Length Note 2			**Standard**	**with AC Adapter**
Dterm Series E Electra Elite	Dterm 8 / 8D	8DLC	300m (984ft)	**Note 3**
		4DLC	300m (984ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	Dterm 16/16D	8DLC	200m (656ft)	**Note 3**
		4DLC	200m (656ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	Dterm 32/32D	8DLC	200m (656ft)	**Note 3**
		4DLC	200m (656ft)	1200m (3937ft)
		2DLC	850m (2789ft)	1200m (3937ft)
	Dterm 8D-1IP	_	Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub	
	Dterm 16D-1IP	_	Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub	
	Dterm 32D-1IP	_	Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub	
	DSS/BLF Console **Note 4**	8DLC	-	300m (984ft)
		4DLC	-	1200m (3937ft)
		2DLC	-	1200m (3937ft)
Operator Position	Attendant Terminal		Same as Dterm Series E	
	SN716 Desk Console	8DLC	300m (984ft)	**Note 3**
		4DLC	350m (1148ft)	1200m (3937ft)
		2DLC	350m (1148ft)	1200m (3937ft)

Note 1: *Loop resistance includes an internal resistance of telephone set or distant exchange.*

Note 2: *Cable length is based on the following conditions.*

- *Diameter of the cable is 0.5 mm.*
- *The Protection arrester is not inserted between the terminal and PBX.*

Note 3: *When using 8DLC card, it is not available for long line function, even if it is equipped with AC Adapter.*

Note 4: *The DSS/BLF Console requires local AC/DC supply.*

Dterm Cordless Terminals

With the NEC Cordless product line, the distance of your handset cord does not govern your work area. The mobility, convenience and reliability of these cost-effective telephones empower employees to answer calls regardless of their location. Potential customers do not waste time playing telephone tag, and employees are not glued to their desk waiting for that all-important call. Business is not lost because of callers being able to reach a live person. The NEC Cordless application is the ideal solution for those businesses that require mobility but do not want to invest in a full-blown wireless solution.

Dterm Headset Cordless

Dterm Cordless II

Dterm Cordless Lite II

Dterm Analog Cordless

Dterm Handset Cordless Terminal

DTERM HEADSET CORDLESS

The analog Dterm Headset Cordless telephone is designed for mobility. Its convenient pocket size allows you to speak and listen in a handsfree mode. Designed and engineered using the 2.4 GHz frequency range, the Dterm Headset Cordless provides clear and secure conversations. The extremely small "handset" measures just 2.13 x 0.61 x 3.39 inches. For added value, the unit comes equipped with two 2.5 mm headsets. In addition, the Dterm Headset Cordless offers such features as a 100-number phone book, a 3-line, 16-character backlit handset display, one-touch dialing and vibration alert mode. With its flexible, unique design and rich feature set, NEC's new hands-free cordless headset is a powerful executive business terminal.

Dterm Headset Cordless Features

- 2.4 GHz Digital Spread Spectrum Technology
- Phonebook Locations (up to 100 numbers in total)
- Trilingual Language Option
- 8-Day Standby Battery Life
- 5-Hour Talk Time
- Tone/Pulse Dialing
- Handset Earpiece and Ringer Volume Control
- 32-Digit Redial / 3 Last Number Redial Locations
- 3-Line, 16-Character Backlit Handset Display
- One Touch Dialing
- Mute Feature
- Flash and Pause
- Find Handset
- Call Timer
- Vibration Alert Mode
- AutoTalk - AutoTalk allows you to answer calls by removing the handset from the base.
- AutoStandby - AutoStandby allows you to hang up by simply returning the handset to the base.
- Random Code - Random Code protects you against misbilled calls which might result from your telephone being activated by other equipment. The Dterm Headset Cordless has digital security which automatically selects one of over 65,000 digital security codes for the handset and base.
- RocketDial - RocketDial is a one-touch speed dial key that automatically dials your most important or frequently called number. The number dialed is a preset number stored by the user.
- Two headsets included: one over-the-head, one over-the-ear
- Beltclip
- Charging Base with an AC Adapter included

Dterm Headset Cordless Specifications

- Size: 2.13 x 0.61 x 3.39 inches
- 2.5mm Headset Connection
- 2.4 GHz Digital Spread Spectrum Technology
- 5-Hour Talk Time
- 8-Day Standby Battery Life
- 3.7V, 650mAh Lithium Ion Battery
- 32-Digit Redial
- 3 Last Number Redial Locations
- 3-Line, 16-Character Backlit Handset Display

DTERM CORDLESS II

Dterm designed for employees who need digital multiline desktop functionality in a cordless handset. The 2x16 line display provides valuable calling information. The Dterm Cordless II is designed to eliminate noise and provides a range of up to 350 feet.

Dterm Cordless II Telephone Features

- Advanced Digital Technology eliminates noise and provides an extended range.
- Direct interface to digital interface port on NEC telephone systems.
- Programmable Function Keys let you utilize features of your telephone system with the Dterm Cordless Multiline Terminal.
- 2x16 Digit LCD displays messages, including caller name and number.*
- Conference calling saves time and improves office communication.
- Compatibility enhances the Electra Professional®, NEAX® 2000 and NEAX 2400 telephone systems.
- Message LED indicates messages in voice mail.
- Headset jack for hands-free operation.
- Hold mode for handling multiple calls.
- Transfer function for switching lines.
- Color- black

Note: Different displays - The Dterm Cordless II Multiline Terminal handset operates with the NEC Electra Professional, NEAX 2000 and NEAX 2400 systems supporting the Dterm Series III/E/i interface. Connection with each system will result in system-specific displays.

**Dterm Cordless II Multiline Terminal will display caller name or number if Caller ID is a feature of the host system.*

Dterm Cordless II Specifications

The Dterm Cordless II complies with FCC and IC parts 15 and 68.

General:

- Frequency Control: Phase Lock Loop
- Modulation: Spread Spectrum
- Operating Temp: 0° to 50° C (32° to 122° F)
- Output Power: 63 mW
- Occupied Bandwidth: 1,600 kHz
- Data Transmission Speed: 688 bps

Base Unit:

- Receive/Transmit Freq.: 902 MHz - 928 MHz
- Power Requirements: 10 Vdc from supplied AC adapter
- Size: 4-1/4 in. W x 7-1/2 in. D x 2-1/4 in. H
- Weight: approximately 15.4 oz.

Handset:

- Receive/Transmit Freq.: 902 MHz - 928 MHz
- Power Requirements: Nickel-Cadmium Battery Pack

Battery Charger:

- Power Requirements: 9V DC from supplied AC adapter
- Size: 2-1/5 in. W x 1-2/3 in. D x 8-2/3 in. H (with antenna)

DTERM CORDLESS LITE II

Dterm designed for employees who need digital multiline desktop functionality in a cordless handset. The 2x16 line display provides valuable calling information. The Dterm Cordless Lite is designed to eliminate noise and provides a range of up to 150 feet. The Dterm Cordless Lite is ideal when the range of the Dterm Cordless II is not required.

Dterm Cordless Lite II Features

- 900 MHz FM with ADPCM
- 2-line, 16-digit LCD Display
- Dterm Cordless Lite II Headset Jack
- Channel Selection Control
- Ringer Volume Control
- Handset Volume Control
- Single Key Access to: Conference, Hold, Transfer, and Mute features
- Four Programmable Keys: F1-F4
- Separate Charging Stand with Spare Battery Charging Capability
- Auto Standby
- AutoTalk
- Silent Alarm
- Out of Range Protection
- Low Battery Protection System
- Key Pad Lock Feature
- Wall Mountable Separate Base Unit
- Wall Mountable Separate Charging Unit
- Easy Installation
- Compact Handset Design
- Use with an NEC Digital Multiline Terminal or as a stand-alone device

Dterm Cordless Lite II Telephone Specifications

General:

- Frequency Control: Phase Lock Loop
- Modulation: 900 MHz FM with ADPMC (digital)
- Operating Temperature: 0° to +50° C (+32° to +122° F)
- Bandwidth: 50 kHz
- Data Transmission Speed: 688 bps

Base Unit:

- Receive/Transmit Frequency: 902 MHz - 928 MHz
- Power Requirements: 10 Vdc from supplied AC adapter
- Size: 4-1/4 in. W x 7-5/8 in. D x 2-1/4 in. H
- Weight: Approximately 13.7 oz.

Handset:

- Receive/Transmit Frequency: 902 MHz - 928 MHz
- Power Requirements: NiMH Battery
- Size: 2 in. W x 1-1/4 in. D x 5-1/2 in. H (with antenna)
- Weight: Approximately 5.2 oz. with battery
- Battery: 700 mAh, 3.6V
- Talk Mode: 6 hours (typical)
- Standby Mode: 5 days (typical)

Battery Charger:

- Power Requirements: 9V DC from supplied AC adapter
- Size: 1-3/8 in. W x 1-1/2 in. D x 2-1/4 in. H

ANALOG CORDLESS

This is the ideal choice for employees that would benefit from mobility but do not require the power and versatility of a digital telephone. The Dterm Analog Cordless offers a highly affordable NEC analog solution and easy access too many system features. This is a 2.4 GHz digital spread spectrum cordless solutions with built in speed dial and dedicated feature buttons offering the benefits of a cordless at an analog price.

Analog Cordless Features

- 2.4 GHz Digital Spread Spectrum
- 30-Channel Autoscan
- 10-Number Memory Dialing
- Desk or Wall Mountable
- Tone/Pulse Dialing
- Handset Volume Control
- 32-Digit Redial
- Page/Find
- AutoTalk -- AutoTalk allows you to answer a call by just removing the handset from the base so you do not have to waste time pushing buttons or flipping switches.
- AutoStandby™

Analog Cordless Specifications

The Analog Cordless complies with FCC parts 15 and 68.

General:

- USOC Jacks: RJ-11C
- Frequency Control: Phase Lock Loop
- Modulation: Direct Sequence Spread Spectrum
- Operating Temp.: 0° - 50° C (32° to 122° F)

Handset:

- Transmit Frequency: 2,416.128 MHz - 2,475.520 MHz
- Receive Frequency: 2,406.912 MHz - 2,466.304 MHz
- Power Requirements: Rechargeable nickel-cadmium battery pack
- Size: 2-1/4 in. W x 1-1/2 in. D x 8-1/2 in. H (with antenna)
- Weight: approximately 8.8 oz with battery
- Battery Capacity: 600 mAh, DC3.6V
- Talk Mode: 6 hours (typical)
- Standby Mode: 4 days (typical)

Base Unit:

- Transmit Frequency: 2,406.912 MHz - 2,466.304 MHz
- Receive Frequency: 2,416.128 MHz - 2,475.520 MHz
- Power Requirements: 9V DC from supplied AC adapter
- Size: 4-1/4 in. W x 7-1/2 in. D x 2-1/4 in. H
- Weight: approximately 15.4 oz.

DTERM HANDSET CORDLESS TERMINAL

The Dterm Handset Cordless operates in the 900 MHz analog spectrum and is compatible with the NEAX 2000 IPS, NEAX 2400, and IPK (2-wire station interface). For the clearest reception possible, the handset automatically selects from one of 40 channels to communicate with its base unit. It is also possible to change channels manually should a conversation become distorted while talking. Up to 40 Dterm Handset Cordless telephones may access the 40 channels simultaneously.

The recommended range or distance between the handset and base unit is 30 - 100 feet, depending upon the environment. This range is ideal for employees who need mobility around their immediate workspace.

Each NEC Dterm Handset Cordless Telephone consists of a base unit, cordless handset, belt clip and a standard rechargeable nickel-cadmium battery that supports 4 hours of talk time or 40 hours of standby. The optional headset may be easily connected to the 2.5mm headset connector on the handset.

Calls may be initiated, transferred and conference using the dial pad of the cordless handset. Users always have the option to perform call-handling functions on the base of their Handset Cordless phone, just as they would on a corded Dterm.

Dterm Handset Cordless Terminal

- Fully modular 900 MHz Analog FM spectrum with Voice scramble
- 40 separate voice channels
- 16 programmable line/feature keys with two-color LED indication
- 8 Dedicated Function Keys
- 4 Softkeys
- 24-character by 3-line Display
- Large Message waiting LED
- Built-in Speakerphone
- Electronic Volume/Tone Controls
- Base Unit
- Belt Clip
- Battery
- RF line cord
- ADA compatibility
- standard nickel-cadmium battery (40 hours of standby talk time)

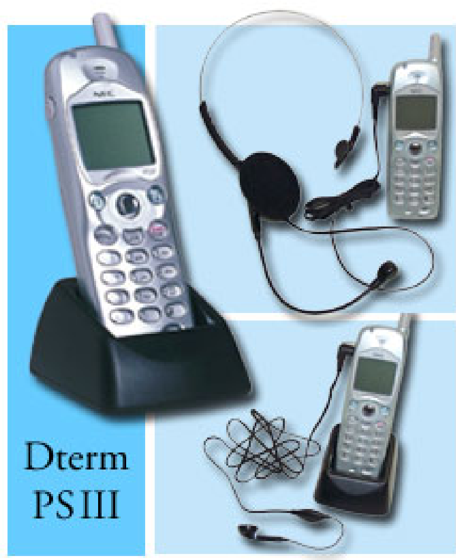
Users always have the option to perform call handling functions on the base of their Cordless Handset phone, just as they would on a corded Dterm.

Wireless Dterm PS Handset

DTERM PS (Personal Station)

The Dterm PS offers portability, clear digital signal quality and has the same features of a desktop phone. Weighing less than 4 oz. with 6 hours of continuous talk time and 300 hours on standby, the Dterm PS phone has been uniquely designed for people on the run. The Dterm PS provides the freedom and convenience of a mobile phone with the advantages and features of a desktop handset.

- Continuous coverage – walk freely around your workplace while on a call.
- No static or fading – audio quality indistinguishable from a desktop phone.
- Small lightweight handset with long battery life.
- Low power operation for compatibility in any environment.
- Up to 512 PS Handsets
- 216 PS Simultaneous Connections



DTERM PS (Personal Station)

PS Handset

Dterm PS Features

- 2-Line Operation with Number Sharing
- Voice Mail Message Waiting
- Calling Party Name and Number display
- Support for Wireless Roaming
- Call Hold & Transfer
- Directory Dial (100 entries with name)
- Speed Dial (20 entries with name)
- Calling Party Number Redial & Last Number Redial (5 entries each)
- Headset Operation with Automatic Answer
- Vibration Alert
- Support for Modem Data
- 2 line, 11 Digit LCD Display (with scrolling)
- User-defined soft keys

And the provisioning of the Dterm PS data port facility to allow software updates protects your investment.

Accessories for the Dterm PS

- Standard Battery Charger Base Unit

- Lithium-ion Battery (provides up to 4 hours of continuous talk-time and 300 hours on standby)

- Leather Cases
  - Leather Case with Swivel Clip - Includes a Swivel Belt Clip with Quick Release
  - Leather Case with Fixed Clip Includes a fixed Belt Clip

  Both cases offer an Integrated D-Ring for use with the optional adjustable necklace**.** The adjustable necklace (from 23" to 42") provides security from dropping the handset while, for example, carrying it in a lab coat pocket.

- Dterm PS Headset
  - Head band type
  - Single ear cover
  - 47" cord

  The Dterm PS Headset is a classic banded headset with a microphone extension. The Dterm PS Headset is typically used in cases of prolonged active use (for example, customer service agents).

- Dterm PS Ear Piece
  - Lower profile Ear Bud receiver
  - In-line Microphone
  - 47" cord

  The Dterm PS Ear Piece is an ear-bud type with an in-line microphone. The Ear Piece is for occasional use or where concealment is desired (for example, hotel security).

- Dterm PS Modem Cable
  - Connects to the modem jack of your laptop or palmtop
  - Supports speeds up to 19.2Kbps
  - Autodial from PC Application (Application must support dialing without waiting for dial tone)

- Dterm PS Assistant Software - The Dterm PS Assistant software and interface cable will allow you to enter the directory and speed-dial entries on your PC and upload to your Dterm PS.

  You can now:
  - Enter, Synchronize and backup your Directory and Speed Dial entries
  - Copy and paste entries from your Desktop Organizer and upload to your Dterm PS.
  - "Clone" all or some of your Dterm PS handset Directory and Speed Dial entries, so everyone in your organization has access to key personnel and department phone numbers.

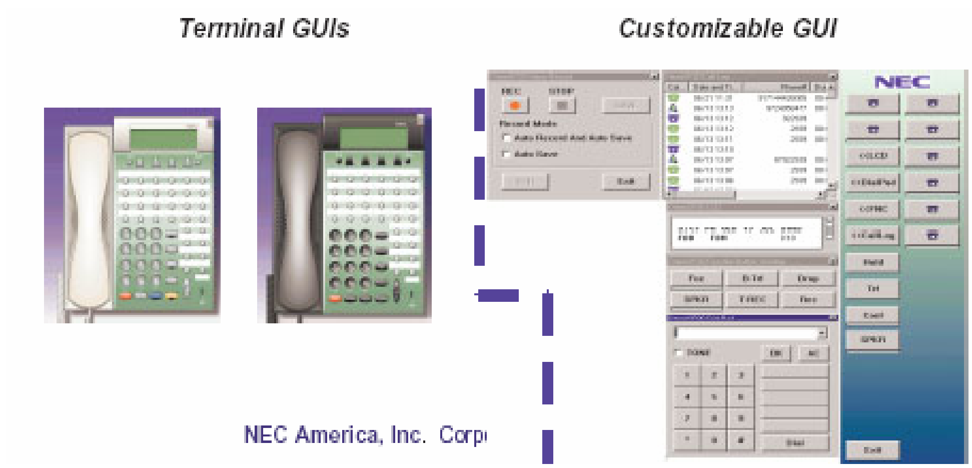
  There are two versions available: Dterm PS Personal Assistant and the Dterm PS Group Assistant. The Personal Assistant supports only one Dterm PS handset while the Group Assistant is designed to manage and maintain Directory and Speed Dial entries for an entire workgroup. The PS Interface Cable connects to the PC serial port and to the Dterm PS interface port, on the bottom of the handset.

Dterm SoftPhone (SP20)

The NEC SP20 with the USB headset allows you to make and receive calls over a VoIP network using a NEC Dterm Series graphical user interface displayed on your laptop. The NEC Dterm SP20 SoftPhone is an ideal solution for employees who are out of the office frequently traveling on business, workers that work both at home and in the office as well as those who require the interaction of data with their work such as a remote ACD agents.

The Dterm SP20 is a full-featured Internet telephone that places calls over a VoIP network using a Dterm series graphical user interface (GUI) displayed on the laptop. It is an ideal solution for employees who work at home, employees who travel frequently, or remote ACD agents who require the interaction of data.

The SP20 SoftPhone has three different types of skin displays. Users have their choice of displays, which look just like the Dterm Series E terminals (in either black or white) as a virtual type of phone, or a customizable graphical user interface. No matter which phone you use as your display, features and functions that existed on the Dterm hardware set are transported to the SP20 upon activation.



Key Features and Benefits

The Dterm SP20 offers a full complement of station features and impressive voice quality over IP for this important segment of mobility and PC-based applications. Most of the features and functions, which exist on the IP Dterm and/or IP enabled Dterm can be instantly transported to the SoftPhone. Basic features offered by the Dterm SP20 include:

Configurable for Peer-to-Peer or Protims IP for IPELC Voice Communications

No need to load new software in order to operate a different protocol. The Dterm SP20 supports both Protims IP and peer-to-peer. Administrators are able to invoke the protocol via the Dterm SP20 control.

Powerful User Interface (Dterm SP20 and Java GUIs) with a NEC Hardware Terminal Appearance

The Dterm SP20 provides a user configurable GUI (Graphical User Interface), enabling the user to access telephone features. Most of the procedures are easy to perform by selecting the buttons on the screen with the mouse. The display may be freely customized in order to create an environment best suited for specific requirements.

Displays call records in Quick Reference List via the Call Log Window

All calls are recorded one by one in the call log. You can find at a glance when and with whom you talked. The Dterm SP20 also offers a call memo function whereby you can record the key points of the call so that you can visually identify the individual records.

Recording Function

Use the personal computer in place of a tape recorder to record the contents of a call. Record the contents of a call in a sound file and reproduce the saved contents of a call anytime from the call log window. Since the file is stored as a .wav file, you can e-mail and forward it to other personnel for their listening pleasure.

Macro Function for Launching or closing Dterm SP20

The macro function will enable the user to launch related applications, which might run simultaneously with the Dterm SP20. One selection and all applications are launched.

Support for multiple algorithms

The Dterm SP20 supports both G.711 and G.729A compression algorithms.

PBX support

The Dterm SP20 is capable of being programmed for either a peer-to-peer or IPELC configuration.

IPELC Software licenses requirements

Software seat requirements for the IPELC configuration are based on a dongle implementation. There is no PBX IP seat licenses required for the Dterm SP20 IPELC operation. The number of licenses depends on the type of licenses key purchased.

The dongles are pre-configured for the following number of licenses:

- 10-seat license
- 20-seat license
- 40-seat license
- 100-seat license

Peer to Peer software licenses requirements

In a peer-to-peer Dterm SP20 configuration, all the software seats are assigned via the PBX. The PBX will require an IP seat license and concurrent seat available in order to activate each softphone.

Dterm SP20 System Requirements

- Protims over IP
- 32IPELC Card and 16VCT Card for compression
- Dterm SP20 Software CD
- External Licensing server via dongle
- NEAX 2000 IPS, 3200 Series software or higher
- SoftPhone 4 seat Licenses : per 4 clients
- 8 Hardware seat licenses: Only required when seat is not available
- Dterm SP20 Software CD

Computer Requirements

- **Computer:** IBM-PC/AT or Compatible
- **Operating System:** Windows XP,2000(SP2),NT4.0 (SP4), ME, 98, 95 English Version
- **Note***: NT4.0 cannot support USB headset*
- **CPU:** Pentium II or greater
- **Memory:** 128Mb or more
- **Hard Disk:** Free Capacity 10Mb or more
- **CD-Rom Drive:** Quad or Faster
- **Mouse:** Window compatible point device
- **Display Resolution:** VGA or higher resolution
- **Network Interface Card:** 10/100base -T
- **Printer Port:** Parallel (Indispensable)
- **USB Port:** USB Headset for Plantronics
- DSP 300 or DSP 400.

Dterm SoftPhone (SP30)

The Dterm SP30 allows customers to capitalize on the advantages of a converged voice and data network whether they're in the office or on the road. The Dterm SP30 combines traditional business communication needs with the data applications your customers require.

The Dterm SP30 optimally delivers high quality voice via a USB-connected headset. With a simple drag and drop, the Dterm SP30 allows telephone dialing from other telephone directory applications such as Microsoft Outlook®, HTML pages and Word® documents, etc. In addition, the Dterm SP30 provides an interface to Microsoft's Telephony Application Programming Interface (TAPI) via NEC OpenWorX integration, allowing TAPI-enabled applications, such as Outlook and ACT, to make and receive calls. The Dterm SP30 has the ability to use a wireless handset (PS) for all voice connections instead of the USB handset. The Dterm SP30 can be displayed in 1 of 4 different colors (black, red, gold and neon blue) in order to reflect the "personality" of the day. The Dterm SP30 also allows for 3 different modes of operation.

Maximized mode: Access to full line of softphone features such as application sharing, member lists, conference mode, chatting capabilities, Internet access and many others are just one click away.

Compact Mode: L-shaped user interface, operating in a small footprint on the PC screen. Compact view allows the softphone to remain active while another application window such as a Word document; database file or email is the primary focus on the PC. With the compact view, the most popular features of the converged softphone are just a click away.

Task Mode: The softphone can be minimized and shown as a task within a Microsoft Operating System. While operating in this mode, the softphone will output an audio notification to the user upon receiving an incoming call. It will be up to the user to utilize the hot key in order to activate the D^{term} SP30 application and answer the call.

The SP30 features three modes of operation:

Maximized mode:
Access to full line of Softphone features.

Compact mode:
L-shaped screen interface offers small footprint.

Task mode:
Minimized icon in system tray.

Key Features and Benefits

The Dterm SP30 offers a full complement of station and converged features for an important segment of mobility and PC-based applications. Even though a majority of the hardware telephone features function on the softphone, the main focus for the softphone is its ability to deliver PC capabilities to the telephone.

Displays call records in Quick Reference List via the Call Log Window: All calls are recorded in a Call Log. Telephone number, date of the call and identification as to whether or not the call was received and answered are all logged. You can find at a glance when and with whom you talked. The Dterm SP30 also offers a call memo function whereby you can record the key points of the call so that you can visually identify the individual records.

Links with PC applications: Now you can collaborate on a white board application (Word® or Excel® document or any other application) that is operating on your PC and the PC at a distant site. Simply place a call to the far-end Dterm SP30 user and activate the application you will be collaborating on. You are now able to have a more productive conversation.

Internet Link: The Dterm SP30 can be assigned with a common database link for fast access to a particular site. This link could be an Internet link that needs to be accessed when receiving or placing a call (i.e., you receive a call from a customer and need to look up information that is contained in a database). Previously, you needed to locate and launch the application before loading the file. Now with the Dterm SP30, simply select the Internet link and the database file is opened, bringing it all together in one user interface.

Real Time Communication: In addition to providing a voice and data collaboration link, you can also chat with a remote Dterm SP30. This is ideal for the real quick conversation you need with a distant Dterm SP30 user. Maybe you’re on a call and need to get a quick response from a co-worker. There’s no need to put the customer on hold and call the co-worker for an answer. Instead, send a chat message and get your answer in real time. The Dterm SP30 will store all your chat messages in a log so that you can use them for future reference. Additional features of the Dterm SP30: forwarding control selection for different call modes, videoconferencing capabilities, automatic downloading of telephony features to the display and diagnostic capacities for audio problem notification.

Phase 2 Features:

Presence/Status: The presence/status functionality allows the user to confirm a buddy's presence/status with a visual indication (ICON) and text message on the Dterm SP30 GUI. The presence/status information is provided by the OpenWorx servers (LSI) package.

8/6 Party Conference Control: When the trunk conferencing card is configured within the voice switch and the voice trunk channel is configured in the program utility for the Dterm SP30, a user can dynamically setup and save future dated collaboration conferences. All the conferences can be activated immediately upon configuration or saved for a future date upon activation. When the conference is activated, the Dterm SP30, with the help of the Voice sever, places calls to the configured participants based on the number which are user predefined. Conferences do not require users to be Dterm SP30 users for voice only conference calls. Only Dterm SP30 users will utilize additional features like application sharing and messaging during call.

PHS/PCS Collaboration: PHS/PCS collaboration links the wireless handset capabilities to the Dterm SP30 GUI. In this mode, all the voice connections will be directed across the PS infrastructure instead of through Dterm SP30 USB handset. The Dterm SP30 gui will give the indication that it has received a call such as caller ID, ringing information and notification as to the state of the line. The only thing that changes is voice path direction being directed to the PS handset. The Dterm SP30 will provide data collaboration interface. The benefits for this functionality are as follows:

1. User has a more reliable voice connection with the voice server. The PS provides a constant voice connection over that of the IP voice connection.
2. Dterm SP30 user now has a handset which can be ported from place to place without the need for programming call forward functionality of the phone.
3. Now the user can be reached by one number, no matter if call forwarding is set or the user is located next to the Dterm SP30.
4. User now has a physical device which they can use for dialing and answering calls instead of working with mouse's and keyboards.
5. User is no longer locked to their PC for receiving and answer calls. The user is as portable as the in-building wireless network.

Voice Recording: A user will be able record the voice connection and save the wave file on a network storage place of their choice. It is recommended all voice recording be stored on a local hard drive and not a network drive. With the activation of the voice recording of the Dterm SP30, no extra recording equipment other than the Dterm SP30 phase 2 application with a USB headset is required. For those specific regions which require the notification to the remote party that a voice connection is being recorded, the Dterm SP30 provides for the setting of an automatic beep tone in the configuration menus. Beep tone can also be conditioned to send a tone notification at user selectable intervals.

Application Collaboration: The NEC Dterm SP30 phase 2 allows users to share ideas, information and programs in a variety of ways while either in a point-to-point connection or 6/8 party conference mode.

- **Videoconferencing -** The Dterm SP30 audio and videoconferencing feature lets you communicate with anyone on the NEC Network.
  - Share ideas, information, and applications using video and audio
  - Send and receive real-time images using Windows-compatible equipment
  - Allows for broadcasting of the live video to other Dterm SP30 users which might not have video transmission capabilities.
  - Use of a video camera to instantly view items, such as hardware devices, road conditions or even personnel, which are displayed in front of the camera

- **Whiteboard -** The whiteboard lets you collaborate in real time with other Dterm SP30 users via graphic design. With the whiteboard, you can review, create and update graphic information.
  - Manipulate contents by clinking, dragging and dropping information on the white board with a mouse/keyboard.
  - Copy, cut and paste information from any Windows-based application into the whiteboard.
  - Use different-colored pointers to easily differentiate participant's comments.
  - Save the whiteboard contents either at the local side or distant end location
  - Load saved whiteboard pages, enabling you to prepare information before a conference, then drag and drop it into the whiteboard during an audio meeting

- **Chat -** The chat functionality lets you conduct real-time conversations via text with as many Dterm SP30 users as you like. With chat, you can:
  - Type text messages to communicate with other co-workers during a conference
  - All messages are sent in a whisper mode so that they are only received by one party
  - All messages sent and received are saved automatically in the chat log
  - Automatic pop up notification when a chat message is received
  - ICON notification within the chat log identifying different states of the messages

- **File Transfer -** File transfer lets you send one or more files to distant Dterm SP30 users. With file transfer, you can:
  - Send a file to other Dterm SP30 users
  - Accept or reject transferred files

- **Application Sharing -** Dterm SP30 gives you better control over how shared programs are displayed on your desktop and give the person sharing the program control over who uses it.
  - View shared programs in a frame, which makes it easy to distinguish between shared and local applications on your desktop
  - Minimize the shared program frame and do other work if you don not need to work in the current conference program.
  - Easily switch between shared programs using the shared programs taskbar.
  - Approve conference participants' requests to work in the program you introduce.
  - Allow or prevent others from working in a program using the sharing dialog menu.

INASET™

INASET terminals are members of the Dterm IP family. INASET terminals have a Web browser with a large color display and a built-in multi-port Ethernet switch for connectivity to the user's local PC. INASET terminals bring a wealth of information to the desktop, including short text display messages and Web pages specifically tailored for the small screen format

The INASET's basic load includes a graphical telephony application that provides an abundance of telephony information and desktop control that is easy to use with its menu-based interface. Information and controls accessible via the softkeys and feature buttons include:

- **Line status** showing a visual icon display for the status of all assigned phone lines and DSS lines.
- **Caller information** showing a visual text display for things such as time, date and call status information.
- **Telephony Directory** for storing, searching and dialing different profiles which you can categorize and store in one of three different groups: corporate, personal and group.
- **Web access** providing browsing capabilities to display HTML web-based information located on the Internet or Intranet. Also includes support for Java applets.
- **Virtual keys** providing access to features, functions and recent keys activated on the terminal. The user can program display and functions for how they see fit with limited or no administrative support necessary.

By converging a company's voice and data networks, there is only one network to manage. The INASET includes a built-in switch, so you can use a single Ethernet switch port for the computer (data) and the INASET. Because it's an IP based telephone, it can be installed anywhere on a corporate IP network. The INASET is Dynamic Host Configuration Protocol (DHCP) compatible and doesn't require the need to be co-located with the NEAX PBX equipment. INASET supports G.711, G.729a and G.723.1 audio compression for low-bandwidth requirements.

INASET Applications

The INASET is specifically suited for the enterprise environment, including the following end users: managers, purchasing agents, consultants and call agents. But this is just the beginning. This advanced IP business terminal can be programmed to do much more than a standard business phone. For example, the customizable user interface (that's developed with the SDK) can replace single line phones that are typically installed in cafeterias, break rooms, lobbies and manufacturing floors. The INASET's programmability is also ideal for traders, stockbrokers, real estate brokers, executives, PR professionals and any other occupation in which professionals use Internet-related information to interact with the public.

- **Personalization at the desktop:** Individual customization can bring flexibility to the desktop. Software developers can enhance traditional features such as Answer, Redial, Conference, Recall and Help menus.
- **Centralized services:** Information or features that need to be accessed by numerous individuals within the organization can be centralized on servers. Directory service functions, Intranet web site information, customer records, CTI applications and other work group applications are examples of centralized services that could improve portability and use throughout the enterprise network.

INASET terminals

Description	Remarks
ITR-LC-1 (BK)	This INASET terminal has a full-color, 3.75” x 4.75” LCD and 16 programmable keys. Does not support Power over Ethernet (POE). Available in black only.
ITR-240G-1 (BK)	The INASET 240G has a 3” LCD with a gray scale 240 x 160 pixel display and 16 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only.
ITR-320C-1 (BK)	The INASET 320C has a 5.1” color display and 32 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only.
ITR-320G-1 (BK)	The INASET 320G has a 5.1” LCD with gray scale display and 32 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol). Available in black only

INASET Accessories

Description	Remarks
WM-RL UNIT	Wall mount for the ITR-LC-1
INASET Headset Cord 12’	12 Foot Black Handset Cord
INASET Headset Cord 25’	25 Foot Black Handset Cord
AC-R	AC adapter for the INASET
ILPA-R	802.3af power Dongle for INASET Original
ADA-2R	Audio recording adapter for IP terminals
MIC-R (BK/WH) UNIT	External Microphone for INASET 240, 320s
PSA-R	Analog survivable adapter

INASET Basic Network Capability

- Dual port 10/100 Mbps Ethernet switch
- Internal voice packet prioritization
- VLAN support (802.1q & 802.1p frame tagging)
- Configurable ToS bits (DiffServ and IP Precedence support)
- Powering
- Inline power (unused pairs) or
- Direct powering with external wall-plug adapter

PBX System Requirements

- NEAX 2000 IPS R6.1or higher
- IPPAD Card
- 8 Seat Licenses: Supports up to 8 INASET terminals

Chapter 4 Equipment Description

Modules

Abbrev	Name Code	Remarks
SN1617 PIMMD	PIM	Port Interface Module (PIM) Maximum 64 physical ports per PIM. Houses two batteries for protection from short power interruption (for 30 minutes). At maximum configuration, the system consists of eight PIMs, and provides a total of 512 physical ports (64 ports × 8).
SN1658 PIMMF	PIM	Port Interface Module (PIM) for Backup CPU System - Maximum 64 physical ports per PIM. Houses two batteries for protection from short power interruption (for 30 minutes). One PIM is required per Backup CPU System.
SN1480 PIMAF	PIM (PIM0)	Port Interface Module (PIM) for DC–48 V Power Input System Mounting the PZ-PW135 Card. One PIM is required per DC–48 V Power System.
COVER PARTS ASSEM-A	COVER PARTS ASSEM	Cover Parts Assembly One cover parts assembly is required for each PIM.
SN1545 BASERE	BASE/TOP ASSEM	Base/Top Cover Assembly One base and top cover assembly is required for each stack.
TOP COVER ASSEM		
SN1685 BASEUC	BASE	Base for DC–48 V Power Input System Installation Cable to Terminal Blocks from this Base is to be local provided. Use to joint the A361 PIM-DC.
SN1619 BATTMB	BATTM	Battery Module for housing PIM or CS (ZT) backup batteries Houses two pairs of batteries for protection from long power interruption (for 3 hours).

Installation Hardware

Equipment Name	Function
HANGER ASSEM (UL)	Wall Hanger Assembly One HANGER ASSEM is required per PIM for Wall Mounting installation.
19 INCH RACK BRACKET (A)	19-inch Rack Mounting Bracket Type A One bracket is required for one PIM configuration. One bracket is required for the top PIM of multiple module configuration.
19 INCH RACK BRACKET (B)	19-inch Rack Mounting Bracket Type B One bracket is required for the bottom module of multiple module configuration.
MOUNTING BRACKET	Safety Mounting Bracket Used as an overhead hanger for Floor Standing Installation. Wire, chain or eyebolts are to be locally provided, to secure the bracket. To be installed on the top PIM in four or more modules of stack. It provides 1.1G shockproof construction.
I/F BRACKET ASSEM	Inter Frame Bracket Assembly Used to joint the frames in two-frame configuration, for Floor Standing Installation.
BASE TRAY ASSEM	Base Tray Assembly One BASE TRAY is required per one frame for Floor Standing Installation of Stationary Equipment.

System Power

Abbrev	Name Code	Remarks
PZ-PW121	AC/DC PWR	Main Power Supply Card: Input: AC120 V/240 V (50 Hz/60 Hz) Output: –27 V (4.4 A), +5 V (7.2 A), CR (38 mA), +90 V (80 mA) One card is pre-installed per PIM.
PZ-PW122	DC/DC PWR	Power Supply Card for Cell Station (Zone Transceiver): Input: –24 V DC Output: –48 V DC (1.7 A) One card per PIM. A maximum of 16 CS (ZTs) backed up by one card.
PN-PW00	EXTPWR	Power Supply Card for DESKCON: • Provides –48 V DC power. • A maximum of four cards per frame (4PIMs). A maximum of three cards per PIM. • Occupies two physical slots width per card.

Common Control Cards

Abbrev	Name Code	Remarks
MP	SPN-CP24B	Main Processor Card Provides LAN control function, system-based Device Registration Server (DRS), built-in FP, Virtual FP, built-in OAI, built-in SMDR on RS-232C, built-in SMDR on IP, built-in PMS on IP, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. One card is required per system.
MP	SPN-CP27	Main Processor Card for Backup CPU system Provides LAN control function, system-based Device Registration Server (DRS), Virtual FP, built-in OAI, built-in SMDR on RS-232C, built-in SMDR on IP, built-in PMS on IP, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. Two cards are required per Backup CPU system. One card for active MP [MP0] and another card for stand by MP [MP1].
MP	PN-CP31	Main Processor Card for Remote PIM (DMR) Provides LAN control function, System-based Device Registration Server (DRS), built-in FP, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), one RS-232C port, 4-line PB receiver, internal Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS. One card is required per system.
ETHER	PZ-M606-A	Ethernet Control Card Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. Provides Auto Negotiation function. • Set to ON/OFF by the office data setting [For Series 3400 software or later] 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.
FP	PN-CP15	Firmware Processor Card Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three PIMs or more, one each of this card is mounted respectively in PIM2, PIM4, and PIM6.

Line/Trunk (LT) Cards

The following table shows a summary of the Line/Trunk (LT) cards for NEAX 2000 IPS. The LT cards may be installed in slot 00 to 11 of PIM 0-7, with the following conditions.

- Maximum 64 ports per PIM

Abbrev	Name Code	Ports	Remarks
LC	PN-8LCAA	8	8-line Analog Line Circuit Card for Single Line Telephones Loop resistance : Maximum 600 Ω Provides Message Waiting Lamp control, momentary open, Line Test functions for each circuit.
	PN-4LCD-A	4	4-line Analog Line Circuit Card for Single Line Telephones Loop resistance : Maximum 600 Ω Provides momentary open and Message Waiting Lamp control, Line Test functions for each circuit. Equipped with +80 V DC-DC on-board power supply.
	PN-8LCAD	8	8-line Analog Long Line Circuit Card for Single Line Telephones; Provides Caller ID Display on analog station. Loop resistance: Maximum 600 Ω Provides Message Waiting Lamp control, momentary open functions for each circuit.
	PN-4LLCB	4	4-line Analog Long Line Circuit Card for Single Line Telephones; Provides Caller ID Display on analog station. Loop resistance for PB/DP type: PB : Maximum 1200 Ω DP (20 PPS) : Maximum 1700 Ω DP (10 PPS) : Maximum 2500 Ω Including the internal resistance of the distant office equipment Provides Message Waiting Lamp control, momentary open/reverse functions for each circuit. PZ-PW122 is required.
	PN-AUCA	2	2-Line Long-Line card provided with the Power Failure Transfer (PFT) function. Used as 2-Line LLC with PFT or 2-Line DID Loop Resistance: MAX.2500ohms (LLC) Two Reverse Circuits Included On-Board Power (-48VDC) included

Line/Trunk (LT) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
DLC	PN-8DLCP	8	8-line Digital Line Circuit Card for Dterm Series *i*/75/65 (Series E/III), DSS Console, ATTCON, DESKCON [–27 V version, 2-wire type]
	PN-8DLCL	8	8-line Digital Line Circuit Card for Dterm Series *i*/75/65 (Series E/III), Dterm Series *i*/70/60 (IPK/Elite/Electra Pro), DSS Console, ATTCON, DESKCON [–27 V version, 2-wire type]
	PN-4DLCQ	4	4-line Digital Line Circuit Card for Dterm Series *i*/75/65 (Series E/III), DSS Console, ATTCON, DESKCON [–27 V version, 2-wire type] Provides Line Test function.
	PN-4DLCM	4	4-line Digital Line Circuit Card for Dterm Series *i*/75/65 (Series E/III), Dterm Series *i*/70/60 (IPK/Elite/Electra Pro), DSS Console, ATTCON, DESKCON [–27 V version, 2-wire type] Provides Line Test function.
	PN-2DLCN	2	4-line Digital Line Circuit Card for Dterm Series *i*/75/65 (Series E/III), Dterm Series *i*/70/60 (IPK/Elite/Electra Pro), DSS Console, ATTCON, DESKCON [–27 V version, 2-wire type] Provides Line Test function.
ILC	SPN-2ILCA	8	2-line ISDN Line Circuit Card Provides a physical interface to ISDN Terminals. Occupies 8 time slots per one card.
COT	PN-8COTS	8	8-line Central Office Trunk Card (Loop Start/Ground Start Trunk) Provides loop detection, sending/detecting ground on Tip/Ring wire.
	PN-8COTQ	8	8-line Central Office Trunk Card (Loop Start Trunk) Provides loop detection, receiving/sending the Caller ID (CLASS SM) signal. μ-law only.
	PN-4COTB	4	4-line Central Office Trunk Card (Loop Start/Ground Start Trunk) Provides loop detection, sending/detecting ground on Tip/Ring wire.
	PN-4COTG	4	4-line Central Office Trunk Card (Loop Start trunk) Provides loop detection, receiving/sending the Caller ID (CLASS SM) signal. μ-law/A-law.

Line/Trunk (LT) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
DID	PN-4DITB	4	4-line Direct Inward Dialing Trunk Card Provides loop detection, sending reverse signal and PB to DP signal conversion. Equipped with –48 V DC-DC on-board power supply.
ODT	PN-2ODTA	2	2-line Out Band Dialing Trunk Card Used as either a 2-wire E&M trunk or a 4-wire E&M trunk. Equipped with –48 V DC-DC on-board power supply. Both No. 0 and No. 1 circuits must be set to the same purpose (2-wire or 4-wire) in one card.
	PN-4ODTA		4-Line Out Band Dialing Trunk Card Used as either a 2-wire E&M trunk or a 4-wire E&M trunk. Maximum 4 cards per PIM. When it accommodates in the LT08 to LT11 slots of the PIM, connect speech line circuits 2 and 3 with the CN1 connector on the front side of the card.
IP PAD	PN-8IPLA		8-channel IP-PAD Card Provides LAN Interface, Packet Assembly/Disassembly to accommodate Legacy Line/Trunk interface. And provides voice compression DSP control functions such as voice compression control, DTMF relay and FAX relay. • Voice compression protocols: G.711 (64 kbps), G.723.1 (5.3 Kbps/6.3 Kbps), G.729a (8 Kbps) • FAX protocol: Pass-through (G.711, G.726) Provides Auto Negotiation function. • Set to ON/OFF by the switch setting When mounting 24DSP (PZ-24IPLA) card, this card can provides up to 32-channel of IP-PAD (When using G.723.1, provides up to 24-channel of IP-PAD). Two cards can be accommodated per built-in FP/FP card, maximum 8 per system. 10BASE-T/100BASE-TX twisted pair cable is connected directly to this card.
	PZ-24IPLA		24-channel DSP Card for 8-channel IP-PAD Card Provides Packet Assembly/Disassembly to accommodate Legacy Line/Trunk Interface. And Provides voice compression DSP control functions such as voice compression control, DTMF relay same as IP-PAD (PN-8IPLA) card. Used to expand the IP-PAD channel up to 32-channels.

Line/Trunk (LT) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
8RST	PN-8RSTG	8	8-line PB Receiver Card Used for a PB station line, DID or tie line.
4RST	PN-4RSTF	4	4-line Sender Card for Caller ID Display on analog Single Line Telephones PN-4LLCB is required.
CFT	PN-CFTB	10	6/10 Party Conference Trunk Card Use of one card: Can control a conference of up to six participants. Use of two cards: Can control a conference of up to ten participants.
DAT	PN-4DATC	8	4-line Digital Announcement Trunk Card Recording duration: Maximum 120 seconds Occupies 8 time slots per one card.
	SPN-2DATA	4	2 Channel Digital Announcement Trunk (60 seconds per Channel)
DK00	PN-DK00	0	8-circuit External Relay Control/External Key Scan Card. Provides the above-mentioned control functions on a per circuit basis.
TNT	PN-TNTA	4	2-line Tone/Music Source Interface Card Used for BGM or Music on Hold. Provides two jacks for an external tone/music source.
VCT	SPN-4VCTI-B	4	4-channel CODEC Card for IP Trunk Card Provides voice compression DSP control functions such as voice compression control, DTMF relay and FAX relay. • Voice compression protocols: G.711 (64 Kbps), G.723.1 (5.3 Kbps/6.3 Kbps), G.729a (8 Kbps) • FAX protocol: T.30 Used together with IP Trunk (PN-IPTB) card. Four cards can be accommodated per IP Trunk (PN-IPTB) card, maximum 32 per system.
	SPN-4VCTI-B (H.323)	4	4-channel voice translator CODEC card for H.323 includes BUS cable. Voice compression: G729A /G723 /G711. Up to qty 3 are used with one SPN-IPTB-B (H323) to provide 12 IP Trunk Channels. Max 3 VCT's per each IPT, max 32 VCT's per system. VCT cards connect together and to IPT via front connected bus cable. One cable is included with each VCT card.

Line/Trunk (LT) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
CSI	SPN-2CSIA	8	2-line Zone Transceiver Interface Card Used to interface with the ZT, based on S-interface. Maximum two ZTs can be connected per CSI card. Occupies 8 time slots per one card.
	SPN-4CSIA	16	4-line Zone Transceiver Interface Card Used to interface with the ZT, based on U-interface. Maximum four ZTs can be connected per CSI card. Occupies 16 time slots per one card.
Router	PN-RTA	0	In-Skin Router Card 10/100BASE-TX: 1, 10BASE-T: 1, RS-232C (D-sub 9pin)
	PZ-M649	0	T1 Digital Trunk Interface (1.5 Mbps) Card Mounted on PN-RTA Card T1 Digital Trunk Interface: 1 Built-in CSU
	PZ-M623	0	Ether Control Card Mounted on PN-RTA Card 10BASE-T: 1
Optical Modem	PN-M10	0	Optical Fiber Interface Card Provides optical fiber interface for T1/E1 Digital Trunk Interface (1.5 M/2 Mbps) Line length : 10 km (6.2 miles) or less Line coding : CMI
PFT	PZ-8PFTB	0	8-line Power Failure Transfer Card To be mounted in PFT slot of PIM. One card per PIM.
	PZ-4PFTA	0	4 Circuit Power Fail Transfer (PFT). Mounts in each IPS DM PIMMF (includes mounting hardware and POWER CA-PFT.)

Line/Trunk (LT) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
VM00	PZ-VM00-M	4	4-port Voice Mail Card (NEAXMail AD-8) One card per system. Number of ports: 4 ports (Up to 8 ports when PZ-VM01 is mounted). Occupies three physical slots width per card. To be mounted in LT00 slot of PIM.
VM01	PZ-VM01	4	4-port Voice Mail Extension Card To be mounted on PZ-VM00/VM00-M.
VM02	PZ-VM02	4	This card consists of a digital signal processor for port interface (4 ports), central processor unit for controlling various data, hard disk unit to read/write the application software, and an internal modem (14.4 Kbps) for remote maintenance. Moreover, mounted into the LT00 slot (for CPU card) of the PIM0. One card per PBX is available.
VM03	PZ-VM03-M	4	This card consists of a digital signal processor for port interface (4 ports), central processor unit for controlling various data, hard disk unit to read/write the voice mail application program and voice mail information, and an internal modem (14.4 Kbps) for remote maintenance. Moreover, this card can provide 16 ports digital line circuit interface, and is mounted into the LT00 slot (for CPU card) and LT01 slot (for DSP card) of the PIM0. One card per PBX is available.
VM04	PZ-VM04	4	This card provides additional 4 ports for transmitting/receiving the voice information, and is used for expanding the port interface up to 12 ports. It is mounted on the VM03 card.
VM05	PZ-VM05	4	This card provides additional 4 ports for transmitting/receiving the voice information, and is used for expanding the port interface up to 8/16 ports. Moreover, this card is used for expanding the fax port interface up to 4 ports. It is mounted on the VM03/VM04 card.
VM06	PZ-VM06	4	This card provides additional 4 ports for transmitting/receiving the voice information, and is used for expanding the port interface up to 8/16 ports. It is mounted on the VM03/VM04 card.

Application Processor (AP) Cards

The AP cards may be installed in slot 00 to 11 of PIM 0-7, with the following conditions:
- Maximum 24 cards per system
- Maximum 256 ports per system

Abbrev	Name Code	Ports	Remarks
DTI	SPN-24DTAC-B	24	1.5M Digital Trunk Interface Uses 24 Application Time Slots Max of 240 DTI Ports, AP Number 4-15 & 20-31 Lower AP Hwy and Upper AP Hwy. Retrofit to use upper highway must mount in PIM 0 Applications; Max 10 T1 1.5M AMI (24ch), Max 8 ISDN-PRI (23B+D) w/ Max 8 SPN-SC01 DCH-C (AP), Max 8 CCIS (23B+C) w/ Max 8 SPN-SC00 CCH-D (AP). -DTE Jitter Transfer, -Fuse capacity (500mA), Built-In CSU.
	SPN-30DTC-UA	31	2M Digital Trunk Interface Uses 31 Application Time Slots. Max of 120 DTI Ports w/this Interface. AP Number 4-15 & 20-31 Lower AP Hwy Applications; 2M AMI (30ch) T1, CCIS (30B+C) requires SPN-SC00, CCH-D (AP) for each CCIS Link.
PRT	SPN-24PRTA-C	25	Digital Trunk/ D-Channel Handler (T1/DCH) Provides ISDN Name Display NI-2. Combination Card includes T1 and DCH. 1.5M Digital Trunk Interface. Uses 25 Application Time Slots. Max 8 ISDN PRI w/this Interface AP Number 4-15 & 20-31 Lower AP Hwy and Upper AP Hwy. Applications; ISDN-PRI (23B+D), -DTE Jitter Transfer, -Fuse capacity (500mA), Built-In CSU.
CCT	SPN-24CCTA-A	25	Digital Trunk/ C-Channel Handler (T1/CCH) Combination Card includes T1 and CCH, 1.5M Digital Trunk Interface. Uses 25 Application Time Slots. Max 8 CCIS Links with this Interface. AP Number 4-15 & 20-31 Lower AP Hwy and Upper AP Hwy. Applications; CCIS with Point to Point T1.
BRT	SPN-BRTC	4	1 Circuit BRI Trunk National ISDN-Uses 2 Application, Time Slots AP Number 4-15 & 20-31 Lower AP Hwy
	SPN-4BRTA-D	8	4 circuit Basic Rate (2B+D) Interface Trunk Card Accommodates four two channel PCM digital lines. Both Point to Point and Point to Multi-Point are supported.

Application Processor (AP) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
CCH	SPN-SC00 CCH-D	1	Common Channel Handler for CCIS 1 per Destination-Site (Max 8 per sys) Required for Event Based CCIS, T1 Drop/Insert CCIS w/external DSU or Fractional T1-CCIS. Event Based CCIS requires using ISDN PRI or BRI trunks with the CCH-D. T1 CCIS requires using a 24DTA with the CCHD. 1 AP Time Slot with out Central Billing. 1 to 9 APTS dependant on Central Billing Dependant on Position in Network. Center Office, 2 APTS per CCH Link. Tandem Office, 2 APTS /CCH. Link + (# of CCH Links-1), End Office, 1 Link Only, 2 AP Time Slots. AP Number 4-15 & 20-31 Lower AP Hwy
DCH	SPN-SC01 DCH-C	1	D-Channel Handler for ISDN PRI Uses 1 Application Time Slot\AP Number 4-15 & 20-31 Lower AP Hwy. Requires using 24DTA and DCH-C.
ICH	SPN-SC03B 8ICH	4	D-Channel Handler (8ICH) with Flash ROM Uses 4 Application Time Slots, Required for BRI Station Side (One SC03 for every four PN-2ILCA cards) AP Number 4-12, Lower AP Hwy
CSH	SPN-SC03B 8CSH-C	4	Wireless ZT Handler (CSH) with Flash ROM Supports both S-Interface and U-Interface. Supports Short Text Message and Dukane Nurse Call system. Uses 4 Application Time Slots. IPS can use PN-4CSIA-A (U-Interface) or SPN-2CSIA. (S-Interface). Retrofit must use SPN-2CSIA. (S-Interface) only. This card is required to expand the number of PS from 256 to 512.
SIG	SPN-SC01 QSIG	1	QSIG Protocol Handler Basic Calling Supported (Receive, Originate) Supports I /A /W ETSI, ETS-300-172; Uses 1 Application Time Slot. AP Number 4-15 & 20-31 Lower AP Hwy Used with SPN-24DTAC-B (AP)
	SPN-24PRTA-QSIG (AP)	25	QSIG combo card with Protocol Handler and 24DTAC. Supports the following Supplementary Services (SS): SS-CLIP, SS-CLOP, SS-CLIR, SS-CONP, SS-CNIR and SS-CONR. Implementation of ID-SS (ETS 300 173) and Name ID-SS (ETS 300 238) is based on basic specifications of ETS 300 172 by ETSI at the Q reference point between PBX’s

Application Processor (AP) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
AP00	SPN-AP00B MRC-C	2	Provides four RS-232C ports and is used for Call Accounting (SMDR) both 2400 and 1400 formats, Property Management System (PMS), Hotel-Printer and Message Center Interface (MCI), CCIS Centralized SMDR/MCI; PZ-M537 expands Buffer (1600 - 27000). RS-232C: Synchronous One Port, Asynchronous Three Ports. AP Number 4-15 & 20-31 Lower AP Hwy
	SPN-AP00B MRC-E	2	Application Processor card. – One AP00 card per system. Supports “DND Automatic Set/Reset at appointed time”, SMDR with only 2400 SMDR format, and MCI. (Does not support 1400 SMDR format, PMS and Hotel Printer).
EX MEM	PZ-M537	0	Expansion Memory for AP00-B
4RST	SPN-4RSTB-B	4	4 Circuit MF Sender for T1 ANI Supports T1 Feature Group D (number only). Uses 4 Application Time slots, AP Number 4-15 & 20-31 Lower AP Hwy. Max: 4-Cards per system including PN-8RSTA/PN-4RSTF/PN-4RSTG and SPN- -4RSTB-911(AP).
	SPN-4RSTB-911	4	4 Circuit MF Receiver for Enhanced 911 Requires CAMA Trunks. Uses 4 Application Time Slots, AP Number 4-15 & 20-31 Lower AP Hwy. Max: 4 - Cards per system including PN-8RSTA/PN-4RSTF/PN-4RSTG and SPN-4RSTB-B (AP).
Caller ID Class	SPN-4RSTC-A SPN-4RSTC	4	4 Circuit FSK Receiver for Analog Trunk CALLER ID Max: 4-Cards including RSTB and RSTB-911AP Number 4-12, Lower AP Hwy, Used with PN-8COTS
IPT	SPN-IPTB-A	0	16 Channel IP Trunk Card. (G.729a. Payload 40ms), VoIP P-P and PMP. Max.1/PIM, Max 8 cards per system, Max 127 channels per system. Has built-in CCH supporting up to 255 nodes. Has one RJ45 connectors for 10/100 Ethernet/Fast Ethernet to router. Up to quantity 4 SPN-4VCT are connected to one SPN-IPT.
	SPN-IPTB-B(AP) (H.323)	0	VoIP H.323 IP Trunk card. Supports up to 12 channels. (G.729a. PAYLOAD 40ms). Also requires CCIS and IPT key per IPTB and NEC GK1000 software. Has one RJ45 connectors for 10/100 Ethernet/Fast Ethernet to router. Up to quantity 3 of SPN-4VCT are connected to one SPN-IPT.

Application Processor (AP) Cards (Cont.)

Abbrev	Name Code	Ports	Remarks
CFT	SPN-CFTC	32	32 Party Digital Conference (Group Call/Meet Me) • Group Call Automatic • Group Call Broad Cast • Group Call Two Way Calling
Wireless Roaming	SPN-AP00B DBM-C	2	Database Memory Card for WCS IP Roaming. - Required for Wireless Roaming and OAI applications using free Location Facility (FLF)", PBX receives a copy of the database from OAI server by FLF function. The PBX stores this database on the SPNAP00B DBM-B card, where it can refer to the data and convert ID codes to station numbers. Expands number of Wireless PS handsets to 512.
	SPN-SC01 DCH-Q	1	DCH Card for Wireless Roaming (Q.931.a) 1 Card per system. Installed With PN-24DTA-A / 30DTC. Uses 1 Application Time Slot. AP Number 4-15 & 20-31 Lower AP Hwy

Chapter 5 Feature Description

Business/Hotel Motel Features

Account Code
This feature, when used with Station Message Detail Recording (SMDR), allows station users and Attendants to enter a cost accounting or client billing code (up to 16 digits) into the system.

Add-On Module
This feature allows the Add-On Module to be combined with a Multiline Terminal when there are insufficient line or trunk keys provided at the Multiline Terminal. When the Add-On Module unit keys are programmed as line/trunk keys, the additional 25 lines/trunks and the existing lines/trunks set for the Multiline Terminal can be accessed directly (maximum of 49 lines/trunks). The station speed dialing function can be assigned for all keys on the Add-On Module unit. Also, one of the last 3 keys can be used as a Day/ Night change key.

Alarm Indications
Faults are indicated by the Major/Minor (MJ/MN) lamps located on the AC/DC Power Supply and, optionally, an external alarm display unit. Station Application not applicable.

Alphanumeric Display
The LCD on Multiline Terminals is used to provide alphanumeric information including clock/calendar and call processing information. Station Application All Multiline Terminals with an LCD display.

Analog Port Adapter
This feature allows an Analog Port Adapter unit combined with a legacy Multiline Terminal to connect to an analog terminal such as an analog telephone, Modem, and PC with built-in Modem. There are two communication modes for the terminal connected via the Analog Port Adapter as shown below:

1. Single Port Mode
   A Multiline Terminal and an analog terminal share the same port. In this mode, the Multiline Terminal and the analog terminal cannot be used simultaneously.
2. Dual Port Mode
   A Multiline Terminal and an analog terminal use different ports. In this mode, the Multiline Terminal and the analog terminal can be used simultaneously.

Announcement Service
This feature allows station users to record messages on Digital Announcement Trunk (DAT) cards. When a station user dials the feature access code for this feature, the user receives the corresponding message from the system. Also Announcement Service can be used to provide a voice message in the following cases;
- An incoming C.O. line/Tie line call has been transferred and encounters a busy or no answer condition
- An incoming DID line/Tie line call has been terminated to a station and encounters a busy or no answer condition
- Internal Recorded Message in place of Music on Hold
- Night Announcement

Answer Key

An Answer Key is provided on all Multiline Terminals. The Answer Key can be used to answer incoming calls on outside lines, and primary or secondary extensions. When the Answer Key is used to answer an incoming call with a call in progress, the first party is placed on hold and the second party is connected. If the Answer Key is depressed while in a three-party call, the user can alternate between each party and a Broker's Call is established.

Attendant Assisted Calling

This feature allows a station user to ask an Attendant for assistance in originating a call. Three methods are available: non-delay, delay, and passing dial tone.

Attendant Camp-on

This feature permits the Attendant to hold an incoming call in a special mode when the desired station for the transfer is busy. The Attendant sends a Camp-On tone to the busy station. When that station becomes idle, it is automatically alerted and connected to the waiting party.

Attendant Console
SN716 DESKCON

The Attendant Console (SN716 DESKCON) operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and reentering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console.

Attendant Called/Calling Name Display

This feature provides a display of the calling/called party's name on the Attendant Console LCD for Attendant Called/Calling Name Display. On attendant-to-station calls, the LCD display the name assigned to the primary extension of the station. On attendant-to-trunk calls, the LCD displays the name assigned to the trunk route of the trunk.

Attendant Called/Calling Number

This feature provides a display of the station number and station name on the Attendant Console during an Attendant-to-station connection. During an Attendant-to-trunk connection, the same display shows the trunk route designation and a trunk identification code (4 digits).

Attendant Call Selection

This feature allows assignment of keys on the Attendant Console to particular types of trunk routes (such as WATS or FX) and particular types of service calls (such as Attendant recalls, intercept calls, etc.). LEDs indicate the type of incoming call and pressing the associated key allows the Attendant to answer the calls in any order.

Attendant Console Lockout - Password

This feature allows the Attendant Console to be set into a lockout mode. This disables the console from originating or receiving calls and setting or resetting service features. To return the Console to its manual operating condition a password is required.

Attendant Do Not Disturb Setup And Cancel

The Attendant has the ability to enter and remove individual stations from Do Not Disturb (DND). Additionally, the Attendant can set one preassigned group of stations into, or out of, Do Not Disturb.

Attendant Interposition Calling/Transfer
This feature allows any Attendant to directly converse with another Attendant and also allows Attendants to transfer calls from their console to another Attendant's console in systems where Multiple Console Operation has been provided.

Attendant Lamp Check
This function is used to check the status of keys, lamps, and LCDs mounted on the Attendant Console to verify that various operations of the Attendant Console are functioning normally. The check is done by a preset procedure.

Attendant Listed Directory Number
This feature provides a display of the Listed Directory Number on the Attendant Console when the operator has answered a Listed Directory Number call.

Attendant Loop Release
This feature allows an Attendant Console loop to become available for a second call as soon as the Attendant has directed the first call to a station, even if that station does not answer.

Attendant Programming
This function is allowed only for the Attendant Console and is used to execute DISA code set up, speed dial programming, and system clock set up operations.

Attendant Training Jacks
The SN716 DeskCon provides two headset/handset jacks on the comnsole, for trining operations. Normal call handling procedures apply. When jacks are used for training, both handsets can be used for listening and talking.

Audible Indication Control
This feature allows the Attendant to adjust the volume of audible indications received at the Attendant Console.

Call Processing Indication
This feature provides visual indications of all calls being processed or awaiting processing at the Attendant Console.

Call Queuing
This feature provides the Attendant the ability to handle a series of exchange network calls in the order of their arrival, (first in, first out) thereby eliminating unnecessary delays.

Call Splitting
This feature allows the Attendant to confer privately with one party on an Attendant handled connection without the other party overhearing.

Call Waiting Display
This feature provides a visual indication to the Attendant when one or more calls are waiting to be answered.

Common Route Indial
This feature allows assignment of incoming DID calls to different Attendant Call Selection keys based on the last 4 digits dialed into the system. Up to eight individual Listed Directory Numbers can be assigned in system programming. When an incoming call to any of these trunks is received, an Attendant Call Selection key will flash and the LCD display will indicate the Listed Directory Number associated with that trunk route.

Dialed Number Identification Service (DNIS)
This feature provides a display of the company name on the Attendant Console when the Attendant has answered a Listed Directory Number or a Tie Line call.

Incoming Call Identification
Incoming calls are identified by various means. Refer to Attendant Called/Calling Number, Attendant Call Selection, Attendant Source Key, Attendant Listed Directory Number and Common Route Indial Features and Specifications.

Individual Trunk Access
The Attendant Console is provided with the ability to access each individual trunk by dialing an associated identification code. This allows detection of faulty trunks during regular testing or after complaints. The Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) has the capability to then busy out the trunk until repair is made.

Multi-Function Key
This feature allows the top row of keys on the Attendant Console to perform and display multiple functions in accordance with the status of call processing.

Multiple Console Operation
This feature allows more than one Attendant Console to operate within the same system.

Pushbutton Calling - Attendant Only
This feature permits an operator to place all calls over Dual-Tone, Multi-Frequency (DTMF) lines from the pushbutton keypad on the Attendant Console.

Serial Call
This feature is activated by the Attendant when an incoming calling party wishes to speak with more than one internal party. When the internal station subsequently disconnects from the Central Office line call, the Central Office party automatically rings back to the same Attendant.

Time Display
This feature provides a digital time display on the Attendant Console LCD. Time is constantly displayed on the Attendant Console LCD. The clock display of the Attendant Console is synchronized with the clock in the sysytem.

Trunk Group Busy Display
A visual indication is supplied to the Attendant when all trunks in a particular trunk group are busy.

Unsupervised Trunk-to-Trunk Transfer By Attendant
This feature allows an Attendant to transfer an incoming or outgoing call on one trunk to an outgoing trunk and exit the connection before the called party answers.

Attendant Delay Announcement
This feature provides an announcement, via a Digital Announcement Trunk Card, to external calls that are not answered by the attendant within a predetermined time.

Attendant Lockout
This feature denies an Attendant the ability to reenter an established trunk or station connection without being recalled by that station after the call is put in consultation hold.

Attendant Overflow
When an incoming call, which has terminated from a trunk to the Attendant Console, remains unanswered after a predetermined time period, this feature provides a change to Night Service for that particular trunk, or an overflow to an outside trunk.

Attendant Override
This feature permits an Attendant to enter a busy connection (station or trunk) using the Attendant Console. When this feature is activated, a warning tone is sent to the connected parties after which, they are connected with the Attendant in a three-way bridge.

Authorization Code
An Authorization Code is a numerical code which will temporarily change a station's Class of Service to a Class of Service assigned to that Authorization Code. This new Class of Service allows access to trunks, dialing patterns, and/or features that would otherwise be restricted.

Automated Attendant
This feature allows the system to answer incoming trunk calls. The system will supply a message and/or dial tone to the caller. The caller can then dial the desired extension number and be directed to that station.

Automatic Call Distribution (ACD)
The Automatic Call Distribution (ACD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the ACD group. Supervisor stations may monitor conversations of agents.

Busy In/Busy Out - ACD
This feature allows an agent in an ACD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the ACD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

Call Waiting Indication - ACD
This feature provides a visual indication when an incoming call to an ACD group is placed in queue, due to an "all agents busy" condition. On external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

Delay Announcement - ACD
This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to an ACD group. A single announcement, or two separate announcements, can be provided.

Hunt Past No Answer - ACD

This feature allows calls targeted at an ACD group to hunt past an agent's station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

Immediate Overflow - ACD

This feature allows a call directed to an ACD group to immediately overflow to another ACD group, upon encountering an "all agents busy" condition.

Priority Queuing - ACD

This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters an ACD queue. When a call is considered as a priority, it is placed at the beginning of the queue.

Queue Size Control - ACD

On incoming DID/Tie line calls, the system can be assigned a threshold which limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

Silent Monitor - ACD

This feature provides the ACD group supervisor with the ability to monitor a call to an ACD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

Automatic Call Distribution (ACD) with Management Information System (MIS)

The Automatic Call Distribution (ACD) with (MIS) provides a management information system to be used in conjunction with the built-in ACD features of the system. The MIS incorporates a supervisor's terminal for real-time monitoring of agent activity, amber and red alarms, and hard-copy summary reports.

Automatic Camp-on

An incoming Direct Inward Termination (DIT) call which has been terminated to a busy station can be Camped-On automatically. When the busy station becomes idle, the station is automatically called and connected to the camped on incoming trunk call.

Automatic Number Identification (ANI)

This feature receives the calling subscriber's number automatically sent from T1 network using MF signaling and displays the calling number on the LCD of a Multiline Terminal and an Attendant Console.

Automatic Recall

This feature works as a timed reminder. When a call remains on Hold, Camp-On or ringing unanswered for a fixed interval after being transferred, the station that initiated the hold, transfer, or Camp-On is automatically alerted.

Automatic Wake-up

This feature allows the system to be programmed to automatically call guest rooms or administration stations at specified times. Upon answering, the guest is connected to a recorded announcement or music source. A printout of unanswered or blocked Automatic Wake-Up attempts for each guest room is provided using the Hotel/Motel printer.

Background Music
Background Music can be provided on a dial-up basis over legacy Multiline Terminal speakers. Incoming voice announcements, ringing and recalls override Background Music. Up to 10 music programs can be offered.

Back Up CPU
NEAX 2000 IPS provides a dual CPU system with two MP cards. When Emergency Notification from hardware is detected, the changeover from an active MP card to a standby MP card will occur. If the active MP card becomes out of order for any reason, the standby MP card starts up automatically. The standby MP card employs a Cold Standby System that will restart initialization by the changeover from the active MP card.

Bandwidth Control
This feature allows assigning an available bandwidth threshold for VoIP Traffic within a location and between locations, and to restrict Outgoing/Incoming Calls when the VoIP Traffic exceeds the threshold. The location is a group of VoIP devices [IP Enabled Dterm, IP PAD, or Peer-to-Peer IP Trunks (built-in IP Trunks)], that have the same VoIP communication parameter (such as CODEC Selection List and ToS field) values assigned. When the VoIP Traffic over CCIS exceeds the threshold, the call can be routed to Legacy Trunks (TDM Network). When exceeding the threshold, the system can store fault information and provide external alarm indication.

Boss / Secretary Calling
A secretary with a Multiline Terminal can use an appearance of the boss' extension to screen calls for that extension, and announce and/or transfer calls to that extension. Additionally, the secretary can call the boss during a busy condition and send a Message Waiting Indication to the boss' station.

Broker's Call
This feature allows a Multiline Terminal or Single Line Telephone user to alternate between two parties, talking to one party while the other party remains on Hold on the same line. The Multiline Terminal user uses the TRF or ANS key to alternate between the two parties. The Single Line Telephone user uses the Hold feature to alternate between the two parties.

Call Back
This feature allows a calling party to set an automatic Call Back when a busy or no answer condition is encountered. When the busy station becomes idle, the station that set the Call Back will be called. In case of Call Back no answer, the Call Back to the setting station is initiated immediately after the called station goes on hook after making a call or accessing a feature.

Call Forwarding

Call Forwarding allows calls directed to a station to be routed to another station, an Attendant, an outside number or voice mail equipment. The types of Call Forwarding provided are:

- Call Forwarding - All Calls
- Call Forwarding - Busy Line
- Call Forwarding - No Answer
- Call Forwarding - Destination
- Multiple Call Forwarding - All Calls
- Multiple Call Forwarding - Busy Line
- Multiple Call Forwarding - No Answer
- Split Call Forwarding - All Calls
- Split Call Forwarding - Busy Line
- Split Call Forwarding - No Answer
- Attendant Call Forwarding Setup and Cancel
- Call Forwarding - Override
- Group Diversion

Attendant Call Forwarding Set-up and Cancel

All of the various types of Call Forwarding can be set up or canceled from both Attendant Consoles.

Call Forwarding - All Calls

This feature allows all calls directed to a particular extension to be rerouted to an alternate destination, regardless of the busy or idle status of the extension. Call Forwarding - All Calls can be set by an Attendant Console, the individual station user, a Multiline Terminal with a secondary appearance of the station's extension, or from another station (which can program itself to be the destination of the rerouting).

Call Forwarding - Busy Line

This feature permits a call to a busy extension to be routed to a pre-designated station, Attendant Console, or voice mail equipment. Call Forwarding - Busy Line can be set or canceled by an Attendant Console, the individual station user, or a Multiline Terminal with a secondary appearance of the station's extension.

Call Forwarding - No Answer

This feature reroutes calls to extensions which do not answer. These calls can be rerouted to another station, an Attendant Console or voice mail equipment. Call Forwarding - No Answer can be set by the individual station user, an Attendant Console, or by a Multiline Terminal with a secondary appearance of the station's extension.

Call Forwarding - Destination

This feature allows a station (A) user to set Call Forwarding - All Calls from another station (B) within the system, to the user's station (A).

Multiple Call Forwarding - All Calls

When a forwarded call is rerouted to a station that has also set a Call Forward, the call can be forwarded to another station. A call can be forwarded up to a maximum of five times, as specified in system programming.

Multiple Call Forwarding - Busy Line

This feature permits a call to a busy station to be forwarded multiple times to a designated idle station.

Multiple Call Forwarding - No Answer

This feature permits a call to an unanswered station, the ability to be forwarded multiple times to a designated station that does not have Call Forwarding - No Answer set or to the Attendant Console.

Split Call Forwarding - All Calls

This feature allows all internal and external calls to a busy extension to be rerouted to different destinations individually, regardless of the busy or idle status of the extension. According to the type of incoming call (Station, C. O. Line, Tie Line, or a call terminated from internal office or via CCIS); Call Forwarding or Split Call Forwarding can be selected.

Split Call Forwarding - Busy Line

This feature allows internal and external calls to a busy extension to be rerouted to separate destinations. Destinations may be an internal station, Attendant Console, or voice mail. And according to the type of a caller (Station/C.O. Line/Tie Line) or a call terminated from internal office or via CCIS, Call Forwarding or Split Call Forwarding can be selected.

Split Call Forwarding - No Answer

This feature allows internal and external calls to a busy extension to be rerouted to separate destinations. Destinations may be an internal station, Attendant Console, or voice mail. And according to the type of a caller (Station/C.O. Line/Tie Line) or a call terminated from internal office or via CCIS, Call Forwarding or Split Call Forwarding can be selected.

Call Forwarding – Logout (Dterm IP)

This feature allows a call terminated to an IP Station in Logout Status to be forwarded to a designated Station, Outside Number, Attendant Console or Digital Announcement Trunk (DAT). This feature is also
applicable to the IP Stations that the LAN Cable is pulled out of or the power is off.

Call Forwarding - Override

This feature allows the call forward destination station to call the station which set call forward. The call forward setting will be ignored.

Group Diversion

This feature allows all calls terminated to an extension that are not answered within a predetermined time to be forwarded to a pre-designated station.

Call Park

This feature enables a station user or attendant to place a call into pre-designated Call Park locations. The station user or attendant is then free to process other calls. This feature is available system wide and for individual tenants.

Call Park - System

When a call is parked by Call Park-System, the call can be retrieved from Call Park by any station in the system.

Call Park - Tenant

When a call is parked by Call Park - Tenant, the call can be retrieved from Call Park-Tenant by any station within the tenant from which the call was originally parked.

Call Pickup

This feature enables a station user to answer any call directed to another station, to a station within the user's own Call Pickup Group, or to a station within a different Call Pickup Group. Three Call Pickup methods are available: Call Pickup - Direct, Call Pickup - Group, and Call Pickup - Designated Group.

Call Pickup - Direct

This method permits a station user to pickup a call to any other station in the system by dialing a specific Call Pickup feature access code and the number of the called extension.

Call Pickup - Group

This method permits a station user to answer any calls directed to other extensions in their preset pickup group by dialing a Call Pickup - Group feature access code.

Call Pickup - Designated Group

This method permits a station user to answer an incoming call directed to another group by dialing the Call Pick-up - Designated Group feature access code and any station within the group to which the ringing station belongs.

Call Redirect

Without answering incoming calls or held calls that terminate to the line keys of a Multiline Terminal, the calls can be transferred to a pre-programmed station or Voice Mail System. Two transferring destination number can be designated per tenant, in system data programming. This feature can be used together with the Caller ID Display feature.

Call Transfer

This feature permits a station user to transfer a call to another station in the system directly, or with assistance from the attendant.

Call Transfer - All Calls

This feature permits a station user to transfer incoming or outgoing calls to another station within the system without attendant assistance.

Call Transfer - Attendant

This feature permits a station user, while connected to an internal or outside call, to signal the Attendant and have the Attendant transfer the call to another station within the system or to an outside connection.

Caller ID Class

This feature receives the calling subscriber’s name and number sent from a public network using a MODEM signal and displays the name or number on an LCD of a Multiline Terminal and Attendant Console.

Caller ID Display
Without answering incoming calls or held calls which terminate to the line keys of a Multiline Terminal, the calling party's information can be confirmed by the indications on the LCD.

Caller ID Station
This feature enables a user to connect analog telephones with Caller ID display function, and provides the calling party’s number and name on the display without answering incoming calls.

Camp-on
This feature provides selected stations or outside calls with Camp-On capability to a busy internal station. Two Camp-On methods are provided. The call waiting method allows a station or an outside party to camp itself on to a busy station. The transfer method allows a transferred outside call to be camped-on to a busy station.

Centrex Compatibility
A combination of features allows full integration of the NEAX2000 IPS with Centrex service.

Check In / Check Out
When this feature is activated, the following operations occur:

- Check In
- Room Cutoff is cleared.
- Check Out
- Room Status printout is supplied.
- Do Not Disturb is reset.
- Room Cutoff is set.
- Message Waiting is reset.
- Automatic Wake Up is cleared.

CID Call Back
When an Incoming Call is terminated from a Station and Trunk with Caller ID (Calling Number Information), and the Called Station does not answer, the Calling Number is registered to the system memory and MW Lamp is lit. After the Calling Number is registered, by operating from a Station, Confirmation, Delete, or Call Back is available. When the Called Station answers, the Calling Number is also registered to the Last Number Redial Memory. The Station can search and call back to that number by the operation of the Last Number Redial/Stack Dial feature.

CID Call Routing
This feature allows designating a call terminating system based on the Calling Party Number received from the network.

Class of Service
This feature permits all stations to be assigned a Class of Service in accordance with the degree of system use desired. The Class of Service is used to assign restrictions for trunk access and feature access.

Code Restriction
This feature allows the NEAX2000 IPS to be programmed to restrict outgoing calls according to specific area and/or Central Office codes. This restriction is controlled on the basis of a three-digit area code or six-digit area and office code numbering plan.

Conference (Three/Four Party)

This feature provides a station user the ability to add-on another party (trunk or station) to a call already in progress. Single Line Telephone users can add up to one additional party and Multiline Terminal users can add up to two additional parties.

Conference (Six/Ten Party)

This feature permits a station user or Attendant (conference leader) to establish a Conference among as many as six or ten parties (including the Conference leader).

Conference (32 Party)

This feature permits a Station User (PS, Multiline Terminal, Single Line Telephone), Attendant, or a Trunk Party to establish a Conference among as many as 32 Parties (including the Conference Leader). Two Conference methods are available: Group Call and Meet-Me Conference.

Group Call

This feature enables a Station User (PS, Multi-line Terminal, Single Line Telephone) within the system or a Trunk Party to establish a Conference among as many as 32 Parties. It also enables a Station User to page a maximum of 31 Parties simultaneously, excluding the Conference Leader. Three Group Call methods are available: Group Call - Automatic Conference, Group Call - Broadcasting, and Group Call - 2-Way Calling.

Group Call - Automatic Conference

This feature enables a Station User to establish a Conference among as many as 32 Parties. From a station or Attendant, a maximum of 31 Stations/Trunks can be paged simultaneously except the Conference Leader. The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

Group Call - Broadcasting

This feature enables a Station User to page a maximum of 31 Parties simultaneously except the Group Call Leader. After Paged Parties answer, the leader can speak to the Paged Parties (the Paged Parties only hear the leader’s voice). The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

Group Call - 2-Way Calling

This feature enables a Station User to page a maximum of 31 Parties simultaneously except the Group Call Leader. After one of the Paged Parties answers, paging becomes 2-Way Calling between the leader and the first answered party and automatically stops paging other parties. The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

Meet-Me Conference

This feature enables Station Users (PS, Multi-line Terminal, Single Line Telephone) within the system, Attendants, or Trunk Parties to join a Conference as many as 32 Parties by dialing a specific Access Code. The Conference participants are automatically connected to the Conference Trunk. Conference participants may call in at preset time or may be directed to do so by a Conference coordinator.

Consecutive Speed Dialing

For Speed Dialing, all digits are registered as a Speed Dialing Code. In the case of Consecutive Speed Dialing, the common portion of the number is registered as a speed calling code, and the remaining digits of each number are dialed by each individual calling station or by using a Station Speed Dial key on a Multiline Terminal.

Consultation Hold

This feature permits a station user to hold any incoming or outgoing CO call, tie line call, or any intra-office call while originating a call to another station user within the system.

Customer Administration Terminal (CAT)

In addition to the Maintenance Administration Terminal (MAT), programming of the NEAX2000 IPS can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming and denying access to others.

Data Line Security

This feature allows line circuits that are used for data transmission to be protected from interruptions such as Attendant Camp-On, Executive Override, and Attendant Override.

Delayed Ringing

This feature enables trunks and station lines to ring immediately at the terminating station, but also, after a programmable period of time has elapsed, to ring at secondary Multiline Terminals with that trunk or line appearance.

Diagnostics

To assist maintenance personnel, the NEAX2000 IPS provides diagnostic capabilities such as fault code generation, device status information and alarm information recording which can be accessed from the Maintenance Administration Terminal (MAT) or Customer Administration Terminal (CAT).

Dial by Name

This feature allows a Multiline Terminal user to search for a desired number by name. The number and name are registered in the system and they are shown on Multiline Terminal LCD. The Multiline Terminal user can search for the desired number by name using up or down soft keys. When the Multiline Terminal user finds the desired number, the call can be originated by pressing the Line/Trunk key or going off hook.

Dial Conversion

The system can be assigned to use rotary Dial Pulse (DP) or Dual Tone Multi-frequency (DTMF) trunks and stations. This feature provides for the repeating of digits dialed by the station user onto the C.O. trunks.

Direct Data Entry

This feature allows a maid or other hotel personnel to enter numeric data to the Property Management System (PMS), using the guest room station for entry through dial operation. The same numerical data can be output to a Hotel/Motel Printer by system data programming.

Direct Digital Interface

This service feature provides the capability to connect trunks from the NEAX2000 IPS directly to T1 carrier links using either a private or public network.

Direct Inward Dialing (DID)

This feature provides for incoming calls from the exchange network (except FX or WATS) to reach any station within the system without attendant assistance.

DID Call Waiting

This feature allows an incoming call on a DID trunk or a tie line to automatically be Camped-On to the destination station if the destination station is busy.

DID Digit Conversion

This feature allows the NEAX2000 IPS to convert the digits received from the serving C.O. to valid station numbers when the C.O. numbering plan differs from the desired station numbering plan.

DID Name Display

This feature allows Name Assignment for a DID Number received from a Public Network, and displays the name on an LCD of a Multi-line Terminal or Attendant Console.

Direct Inward System Access (DISA)

This feature allows an outside caller to access the system using an exchange network connection without Attendant or station assistance. The outside user may originate calls over any or all of the system's facilities such as WATS, FX, Tie Line or CCSA. The outside user can also directly call stations and access miscellaneous trunks for such features as dictation access.

Call Forwarding set by DISA

This feature allows an outside caller to set Call Forwarding – All Calls Direct Inward System Access (DISA) code.

Direct Inward Termination (DIT)

This feature automatically routes incoming network exchange calls directly to a pre-selected station without Attendant assistance. The call can then be processed by the called party. Three-party Conference, Call Transfer, etc., are handled in the same manner as any normal trunk call.

Direct Outward Dialing (DOD)

This feature permits any station user the ability to gain access to the exchange network by dialing an access code and receiving new dial tone. The user may then proceed to dial the desired exchange network number.

Direct Station Selection/Busy Lamp Field (DSS/BLF) Console

This feature allows a Direct Station Selection/Busy Lamp Field (DSS/BLF) Console to be associated with a legacy Multiline Terminal. When the buttons on the DSS/BLF Console unit are programmed for Direct Station Selection (DSS) buttons, up to 60 stations can be directly accessed in addition to those already appearing on the Multiline Terminal. Busy status for each station is indicated by a red LED associated with each button. In addition, the DSS console can provide the following functions:

- Message Waiting - Set/Cancel/Status Display
- Do Not Disturb - Set/Cancel/Status Display
- Automatic Wake Up No Answer - Status Display/Cancel
- Agent Busy Out - UCD - Status Display
- Line Lockout - Status Display
- Room Cutoff - Set/Cancel/Status

Busy Out Status Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Busy Out Status Console. This feature is activated by use of a Function Mode key on a DSS/BLF Console. Busy Out Status for each station is indicated by a red LED associated with each button.

Do Not Disturb Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Do Not Disturb (DND) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. DND set status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel the DND status of other stations using the DND Console.

Message Waiting Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Message Waiting (MW) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Message Waiting status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/reset MW status using the MW Console.

Room Cutoff Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Room Cutoff Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Room Cutoff status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel Room Cutoff to another station using the Room Cutoff Console.

Wake Up No Answer Console

This feature allows an EDW-48-2A unit associated with a Multiline Terminal to be used as a Wake Up No Answer (WU) Console. This feature is activated by a function mode key on a DSS/BLF Console. The No Answer status for each station is indicated by a flashing green LED associated with each button.

Distinctive Ringing

This feature provides Distinctive Ringing patterns to the station so that the station user can distinguish between internal and external incoming calls. This feature also enables the LED associated with the line key of the Multiline Terminal to flash in two colors according to the kind of incoming call.

Do Not Disturb

This feature restricts incoming calls to a station and can be set by an individual station or from the Attendant Console. Placing a station in Do Not Disturb (DND) does not prevent a station from originating a voice or data call or from receiving a data call. This feature also allows a station to ensure privacy from telephone interruptions while on an outgoing call. Additionally, the Attendant Console can place a group of stations in the Do Not Disturb condition.

Do Not Disturb – Group

This feature allows the system to schedule to set/cancel Do Not Disturb for a group of stations at appointed time. The system has up to four patterns of timetable, and each timetable has time to set/cancel Do Not Disturb in a day (the time is programmable by the system). The timetable can be assigned for day of the week. The different timetable can also be assigned for specific dates of the year.

Do Not Disturb - Hotel/Motel

This feature allows the Attendant Console(s), Hotel/Motel Front Desk Instrument(s), guest stations or Property Management System (PMS) terminal(s) to place individual stations into Do Not Disturb. Calls can be placed from stations set in DND.

Do Not Disturb-System

This feature simultaneously restricts incoming calls to a pre-assigned group of stations by operation from the Hotel/ Motel Front Desk Instrument(s). Attendant Console(s) and Hotel/Motel Front Desk Instruments can use the DND OVR key to override this Do Not Disturb setting.

Dterm Assistant

Dterm Assistant is Web-based software which resides on the server and provides end users with the ability to maintain Dterm Multi-line Terminals and the IPS telephony features (such as Speed Dialing) from a Web-enabled PC. The Dterm Assistant operates in a client-server environment and can manage multiple IPS systems over a Local Area Network (LAN)/Wide Area Network (WAN).

Dterm IP

Dterm IP is an IP-based Multiline Terminal which provides a built-in capability of peer-to-peer IP communications. The NEAX2000 IPS system provides the Dterm IP with same IP communications capabilities of an IP Enabled Dterm (The IP Enabled Dterm is a Dterm Multiline Terminal with an add-on IP adapter unit).

Elapsed Call Timer

This feature provides a display of the elapsed time while a Multiline Terminal with LCD is connected to any trunk.

Enhanced 911

This feature allows the PBX to transmit a caller’s emergency service identification information to an Enhanced 911 Emergency system. The 911 notification is also provided to the EMG key of a designated Attendant Console/Dterm.

Executive Calling

This feature allows a station to be assigned a VIP class. This provides special ringing to a called station when that station is idle, and automatic sending of three tone bursts to a called station when that station is busy, provided the call was originated from a station assigned as VIP class.

Executive Override

This feature allows selected users to override a busy condition on a called station. A warning tone is transmitted to both stations in the busy call before the busy condition is overridden, and a three-party Conference is then established.

External Paging with Meet-Me

This feature allows a station user or attendant dial-access to local voice paging equipment and connects both parties automatically after the paged party has answered the page by dialing an access code.

Fax Arrival Indicator

When a call from a C.O. line (Direct-Inward-Termination, Direct-Inward-Dialing, Automated Attendant), station or tie line has terminated to a facsimile machine, a related lamp on a pre-designated Multiline Terminal is caused to light, indicating reception of a facsimile call.

FAX over IP

This feature allows the system to transmit facsimile communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN). Since PBX regards facsimile equipment as one of ordinary telephones, IP Packet Assembler/Disassembler (IPPAD) and Voice Compression Trunk (VCT) are required for facsimile uses over IP network same as legacy stations. The facsimile transmission procedure (T.30 or G.711/G.726 pass-through) is supported with IP-PAD/VCT.

Feature Activation from Secondary Extension

This feature allows the Multiline Terminal user to access an appearance of another extension and program certain features from that extension.

Flexible Line Key Assignment

Multiline Terminals can have any desired line-key assignment. This feature permits assignments to be tailored to each individual's needs. (The terminal's primary extension line appearance is the only line key that cannot be reassigned.)

Flexible Numbering Plan

The NEAX2000 IPS has a Flexible Numbering Plan. All access codes and station numbers and can be assigned in system programming. Refer also to the Single Digit Dialing Features and Specifications, which further increases the flexibility of the system.

Flexible Ringing Assignment

This feature allows lines on Multiline Terminals to be individually programmed to ring or not ring.

Forced Account Code

This feature forces the user to enter an Account Code (up to 8 or 10 digits) for all outgoing calls. The Account Code must be dialed before dialing the outgoing number. Calls are processed only when the dialed Account Codes are valid.

Group Call – Automatic Conference (6/10-Party)

Automatic Conference

This feature allows a Multiline Terminal user or single line telephone user within the system to establish a conference among as many as six or ten parties. From a Multiline Terminal /Single Line Telephone, a maximum of 9 stations can be paged simultaneously plus the originator. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand.

2 Way Calling

This feature allows a Multiline Terminal/Single Line Telephone to page a maximum of fifteen parties simultaneously including the originator. After one of paged parties answers, the paging becomes the 2 Way Calling between the originator and the first answered party, automatically stops paging other parties. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand.

Group Call by Pilot Number Dialing

This feature allows a Station User (Multi-line Terminal/Single Line Telephone/PS) or a Trunk Party to page a group of Stations simultaneously by dialing a Pilot Number. The maximum of 32 Stations can be assigned to a Paging Group, and the Paging Group is associated with the Pilot Number. After one of the Paged Stations answers, the paging becomes a 2-Way calling between the Calling Party and the first answered Station and automatically stops paging other Stations.

Group Listening

When a Multiline Terminal user makes a call using the handset, pressing the SPKR key will allow others to listen through the built-in speaker of the Multiline Terminal. The user may continue talking on the handset at the same time.

Hands-free Answerback

This feature allows the station user to answer a voice call without lifting the handset.

Hands-free Dialing and Monitoring

This feature allows the station user to dial or monitor a call without lifting the handset.

Hold

This feature permits a user to Hold a call in progress. After Hold has been set, the station user can make or answer new calls.

Call Hold

This feature permits a user to Hold a call in progress by sending a hookflash and dialing the Call Hold feature access code, or by pressing the Call Hold key. This line can then be used for originating another call or returning to a previously held call.

Dual Hold

This feature permits a station user who is placed on Hold by another station to place that station on Hold also.

Exclusive Hold

This feature allows a Multiline Terminal user to place a call on Hold and to exclude all other station users from retrieving the held call.

Non-exclusive Hold

This feature allows a Multiline Terminal user to place a call on Hold that may be retrieved by any station that has an appearance of the held line.

Hotel/Motel Attendant Console

The Attendant Console can be programmed to function as a Hotel/Motel Attendant Console. In addition to the business features and functions of the Attendant, the Hotel/Motel Attendant Console can set Room Cutoff (individual and group), Automatic Wake Up, Message Waiting, and Do Not Disturb (individual and group).

Hotel/Motel Front Desk Instrument

A Multiline Terminal with LCD can be programmed to function as a Hotel/Motel (H/M) Front Desk Instrument. This can be used to set and cancel standard H/M features such as Message Waiting, Do Not Disturb, Automatic Wake Up, and Room Cutoff.

Hotline - Inside/Outside
This feature causes the terminal to place a call to another station or to an outside party automatically when the user selects the Hotline extension.

House Phone
This feature allows selected stations to reach the Attendant simply by going off-hook.

Individual Attendant Access
This feature permits a user to call a specific Attendant by dialing an Attendant call code.

Intercept Announcement
This feature provides the automatic interception of Direct Inward Dialing (DID) and Tie Line calls which cannot be completed due to unassigned station or level. The caller hears a recorded Intercept Announcement that informs the caller that an inoperative number was reached, and may supply the number for information.

Intercom
Three types of Intercoms are available: Manual Intercom, Automatic Intercom, and Dial Intercom. Each type of Intercom provides access to a small group of Multiline Terminals with simplified calling methods.

Manual Intercom
The Manual Intercom groups have up to six Multiline Terminals sharing a common signal path. Users can call other members of the Manual Intercom group by pressing a Manual Intercom key; each press sends a tone burst over the speakers of all the terminals in the group. When another user answers the call, a speech path is activated.

Automatic Intercom
Automatic Intercom provides a path for Voice Announcement Calls with Handsfree Answerback between two Multiline Terminals using a line key. Private conversations can be held by using the Multiline Terminal handsets. The Busy/Idle status of the associated Multiline Terminal is displayed on the Automatic Intercom line key LED.

Dial Intercom
Dial Intercom comprises up to 10 Multiline Terminals which can call each other using a dedicated Dial Intercom line key with abbreviated dialing. Dial Intercom calls can be Voice Announce with Handsfree Answerback or ringing calls.

Internal Tone/Voice Signaling
Multiline Terminals can signal incoming internal calls by Voice Announcement or by ringing according to the programmed mode (Voice first or Ring first) of the called terminal. The caller can dial the digit 1 to change from Voice Announcement to Ring Tone or vice versa. The Multiline Terminal assigned this feature can program the following two modes:

- Voice Mode: allows an incoming call to terminate with Voice Announcement.
- Tone Mode: allows an incoming call to terminate with ringing.

Internal Zone Paging with Meet-Me
This feature allows the Attendant Console and system users to page over the built-in speakers of the Multiline Terminals within the assigned zone or all zones.

IP Enabled Dterm

This feature provides a Dterm Series E/Series i terminal, if equipped with an IP adapter unit with a capability to provide a converged infrastructure at the desktop, with a 10Base-T/100Base-TX Ethernet connection to corporate Local Area Networks (LAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm and CCIS network (IP based) on a peer-to-peer connection basis. And, the IP Enabled Dterm also communicates with legacy stations and trunks (TDM based) via IPPAD (IP Packet Assembler/Disassembler). The IP Enabled Dterm provides users with all features currently available in Dterm Series E/Series i terminals.

Last Number Redial

This feature allows users to redial the last station-to-station or outside number they dialed using a feature access key or a feature access code. This is useful when the called station is busy or does not answer.

Least Cost Routing - 3/6 Digit

This service feature allows the NEAX2000 IPS to be programmed to route outgoing calls over the most economical facility (WATS, FX, DDD). Based on the individual area code and office code dialed (6-digit analysis), the system examines the programmed tables and uses the trunk in the order specified.

Line Lockout

This feature automatically releases a station from the common equipment if the station remains off-hook for longer than a programmed interval before dialing. Howler tone may be programmed to be sent to the station in Line Lockout.

Line Pre-selection

This feature provides the station user with two ways to select an idle, held, recalling, or ringing line before going off-hook.

Maid Status

This feature allows the Hotel/Motel (H/M) Front Desk Instrument, Property Management System (PMS) terminal, or guest room station (using special access code) to register the condition of each guest room.

Maintenance Administration Terminal (MAT)

The Maintenance Administration Terminal (MAT) is a personal computer that provides an interface to the PBX via the system’s CPU card. The MAT PC must have the MATWorX program properly installed to communicate with the PBX. MATWorX is required for system software registration and activation. MATWorX is a Graphical User Interface (GUI) program that provides an efficient method to manipulate the PBX database. This program contains extensive help files, Usage Wizards and Tool Tips, with hyperlinks imbedded in the text. The hyperlinks provide quick access to the appropriate Add-In modules. Add-In modules provide a user friendly intuitive method to customize the PBX database.

Message Center Interface (MCI)

This feature provides an interface with a customer supplied Voice Mail System (VMS) that can send Message Waiting lamp control data to the system. The Message Center Interface (MCI) can provide the following operations:

1. When terminating the call to the VMS, the system sends call connection status information to the VMS through the MCI.
2. The VMS sends the Message Waiting Lamp on data to the MCI.
3. The system, upon receiving this control data from the MCI, illuminates the Message Waiting lamp of the corresponding station.
4. The VMS, upon receiving retrieved message information, will send the Message Waiting lamp control data requesting the system to extinguish the Message Waiting lamp of the corresponding station.

Message Registration

This feature provides output from the system to a call accounting system using an RS-232C connector. This allows the Hotel/Motel clerk to retrieve the information needed to charge for local and toll calls.

Message Reminder

This feature allows a user or Attendant to turn on the message waiting (MW) lamp of a Single Line Telephone, or the Message Reminder (MSG) LED of a Multiline Terminal (if assigned).

Message Waiting

Message Waiting – Single Lamp

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, Voice Mail System (VMS) or Property Management System (PMS) terminal to light a lamp (on an uninterrupted or interrupted basis) on a Single Line Telephone or Multiline Terminal to indicate a message is waiting.

Message Waiting – Multiple Lamp

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, Voice Mail Systems (VMS) or Property Management System (PMS) terminal to light multiple line keys on a Multiline Terminal, to indicate a message is waiting. This allows multiple individuals who share the same Multiline Terminal to receive their own Message Waiting indication.

Voice Message Waiting

In addition to the lamp indication control, this feature also provides the Voice Message Waiting service that an originating station user can set the Message Waiting with a recorded message by using the Digital Announcement Trunk (DAT) card.

- Voice Message Waiting – System

  An originating station user can choose the recorded message to be set by dialing the message number associated. The messages are recorded by the predetermined station.

- Voice Message Waiting - Individual

  When setting Message Waiting, an originating station user announces the message to be recorded after dialing the station number.

Miscellaneous Trunk Access

This feature allows the connection of various types of external facilities. In addition to Loop and Ground Start Trunks, the following can also be interfaced with the NEAX2000 IPS: CCSA Lines Code Calling Equipment, Dictation Equipment, Foreign Exchange (FX) Lines, Radio Paging Equipment, and Wide Area Telephone Service (WATS) lines. Refer to separate features, Direct Inward Dialing (DID), and Tie Line Access for more applications of Miscellaneous Trunk Access.

CCSA Access

This feature allows connection to or from a CCSA (Common Control Switching Arrangement) network. A CCSA network is a special, privately-leased network constructed for one customer's exclusive use that replaces or augments the public switched network services. The outgoing connections using CCSA lines are accomplished in the same manner as a normal outgoing call. Incoming calls are received from the CCSA network as a series of digits from the network instead of a ringing signal, and the connection is established in the same manner as a Direct Inward Dial (DID) or Tie Line connection. For Tie Line applications, the customers can construct a network with their own numbering plan. In a CCSA application, the numbers are issued by the C.O. following the CCSA network numbering plan.

Code Calling Equipment Access

Code Calling Equipment consists of external paging units and external dialers requiring dial access from the NEAX2000 IPS.

Dictation Equipment Access

This feature permits dial access to customer provided Dictation Equipment, and in some instances allows them to maintain telephone dial control of normal dictation system features.

Foreign Exchange (FX) Access

An FX line is a line that is extended and terminated at a distant Central Office. With this feature, outgoing calls over the FX line become a local call at the distant C.O.

Radio Paging Equipment Access

This feature provides station users dial access to Radio Paging Equipment, and to selectively tone - or voice/ tone-alert individuals carrying pocket paging devices. The paged party (when on premises) can be connected to the paging party by going to the nearest station and dialing an answer back code.

Wide Area Telephone Service (WATS) Access

This feature allows any station user direct dial access to outgoing WATS lines.

Modem over IP

This feature allows the system to transmit modem communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN).

Multiline Terminal Attendant Position

A Multiline Terminal with LCD can be programmed to function similar to an Attendant position. This Attendant position has limited access to Attendant related features and functions and can be substituted where an Attendant is required but an Attendant Console is not necessary. When a DSS/BLF console unit is associated with this Attendant Multiline Terminal enhanced operation is available.

Music on Hold

This feature plays music when a line is placed on hold. Music is provided by a circuit board memory chip, IP adpter, or a local music source, such as a CD player or a radio.

Night Service

This feature provides a variety of methods for handling incoming calls when the system is in night mode. These include:

- Attendant Night Transfer
- Call Rerouting
- Day/Night Mode Change by Attendant Console
- Day/Night Mode Change by Station Dialing
- Night Connection-Fixed
- Night Connection-Flexible
- Trunk Answer Any Station

Attendant Night Transfer

When the Attendant Console is in Night Service, any operator directed calls (dial 0 calls) are automatically routed to a preprogrammed station. Priority Calls and Off-Hook Alarms which terminate to an Attendant are also routed by this feature.

Call Rerouting

This feature provides flexible reroute capabilities for a variety of calls when the system is in night mode.

Day / Night Mode Change by Attendant Console

This feature provides activation of DAY/NIGHT Mode Change by depressing a predetermined key from the Attendant Console.

Day / Night Mode Change by Station Dialing

This feature allows selected stations to activate a change from day mode to night mode by dialing a special code.

Day/Night Mode Change by System Clock

This feature provides automatic activation of Day/Night Mode Change by using System Clock.

Night Connection - Fixed

This feature allows incoming calls normally terminated to the Attendant to reroute to a predetermined station when the system is placed in Night Service.

Night Connection - Flexible

This feature provides incoming calls normally terminated to the fixed night station to be Call Forwarded to another station.

Trunk Answer Any Station (TAS)

This feature allows any station, other than one with incoming restrictions, to answer incoming calls when the system is in the night mode. When this feature is activated, incoming exchange network calls will activate a common alert signal at the customer premises. By dialing a specified code, any station may answer the call and then extend it to any other station by means of the Call Transfer feature.

Overflow for TAS Queue

If the TAS Call is not answered by predetermined time, the call will be forwarded to predetermined Station/Attendant Console/Announcement Service.

Queue Limit for TAS

When a DID Call is converted to TAS and the number of used Lines reaches queue limit, this feature provides the system to restrict the next call terminating.

Off-hook Alarm

This feature allows a station user to call the Attendant, or a pre-designated station, by simply staying off-hook for a preprogrammed period of time. The calling number is automatically displayed at the Attendant Console, or the pre-designated station if equipped with an LCD.

Off-Premises Extensions

This feature allows the connection of a single line telephone in an off-premises location. The connection to the Off-Premises Extension can be through direct copper or through the local telephone company.

Open Application Interface (OAI)

Provides a computer-to-PBX interface, allowing a computer to control the function of the NEAX2000 IPS. The NEAX2000 IPS can be customized to accommodate most customer applications. Application software can be provided by NECAM, an outside software house, or a customer.

Pad Lock

This feature temporarily restricts telephones from making unauthorized calls by dialing special access code when station users are away from their seats.

Periodic Time Indication Tone

This feature provides a periodic tone to the station user who has made an outgoing call. This feature can be allowed or denied for each station.

Pooled Line Access

A line key can be assigned to access Pooled Lines. Each line key will allow incoming, outgoing, or both-way access to a trunk route.

Power Failure Transfer

This feature provides for specified trunks to be automatically connected to designated Single Line Telephones in the event of AC power loss. It is normally used when the system is not equipped with reserve power.

Priority Call

This feature allows the Attendant to answer a call before other calls, at the Attendant's discretion.

Privacy

This feature restricts Multiline Terminal users from depressing a busy line button and entering a conversation unless permitted by the Multiline Terminal user currently on that line button or if the line button is assigned for Direct Privacy Release.

Direct Privacy Release

This feature allows a station user with a secondary appearance of another extension in the system to access that extension when it is being used by someone else. This feature allows for a simplified method for establishing a conference. In addition, this feature can be used to emulate PC dialing, where a single line extension connected to a PC can appear on a Multiline Terminal and be accessed by the Multiline Terminal user after the PC is completed dialing.

Manual Privacy Release

This feature allows a Multiline Terminal user to enter a conversation on a busy line button if the Multiline Terminal user already in the conversation allows them by releasing Privacy.

Private Lines

Only a C.O. trunk assigned to that specific station is seized when a station user originates an outgoing C.O. call or when an incoming C.O. call is terminated at the station designated by Direct-In-Termination. In this manner, stations and C.O. trunks are to be associated on a 1-to-1 basis.

Property Management System Interface

The NEAX2000 IPS provides a data interface to a locally provided Property Management System (PMS). This enables communication between the NEAX2000 IPS and the PMS in order to provide computer control of Hotel/ Motel features.

Proprietary Multiline Terminal

There are five Multiline Terminals that can be used with the system.

- Dterm Series i
- Dterm Series E
- IPK/Electra Elite

Automatic Idle Return

This feature returns a station to the idle state after 3 seconds of reorder tone is received due to the distant end disconnecting.

Called Station Status Display

This feature provides a display on the status of a called station on the LCD of the calling Multiline Terminal.

Calling Name and Number Display

This feature provides a display on the LCD of the Multiline Terminal receiving a call, indicating the station number or trunk number of the incoming call.

Dynamic Dial Pad

This feature allows to make an outgoing call at first hand by pressing a ten key of Multiline Terminal, without pressing a Speaker key or going off-hook.

Handsfree Unit
The built-in Handsfree Unit enables full Handsfree operation for both internal and external calls (No optional Handsfree Unit is required).

I-Hold / I-Use Indication
Multiline Terminals provide indication of which line keys have been placed on Hold, or are in use by that Multiline Terminal. The LED associated with the line key will give the appropriate indication.

Microphone Control
All Multiline Terminals are equipped with a Microphone Control button with an associated LED.

Multiple Line Operation
This feature allows for the appearance of multiple lines on the Flexible Line Keys and feature keys of all Multiline Terminals.

Mute Key
This feature allows the distant extension user, of a station user that presses a mute key during conversation, not to hear the station user's voice though the station user can hear the distant extension user's voice. By pressing the mute key again, the mute status returns to original conversation.

Prime Line Pickup
This feature allows a Multiline Terminal user to go off hook and originate a call from the line assigned as the Prime Line without depressing the associated line key.

Recall Key
Each Multiline Terminal is equipped with a Recall Key that is used to generate a hookflash to access features provided by the outside exchange, or to abandon a call while retaining the line for origination of another call.

Relay Control Function Key
This feature provides a Multiline Terminal with the ability to activate/deactivate relays (on a PN-DK00) to control external devices.

Ring Frequency Control
The ring frequency of the Multiline Terminal can be controlled on a station basis in system programming (four frequencies are available) or by use of a function key on the Multiline Terminal.

Ringing Line Pickup
This feature provides the ability to answer any call ringing into a Multiline Terminal by just lifting the handset.

Soft Keys

According to the status of the Multiline Terminal, function keys (Soft Keys) are displayed in the third line on the LCD. If the status of Multiline Terminal changes, the Soft Keys will change automatically. Also if the Help key is pressed, explanation of indicated Soft Keys are shown on the LCD.

Volume Control

Multiline Terminals are equipped with common Volume Control keys for:

- Built-in Speaker / Handset Receiver Volume.
- Ring Volume.
- C.O. Transmission Level.
- LCD contrast.
- Ring Tone Frequency

The Volume Control keys are located on the lower front side of Multiline Terminals (UP and DOWN).

Remote Hold

This feature allows a Multiline Terminal user to hold it on the line button of transferred terminal, by pressing the Hold key.

Remote PIM over IP

When IPSDMR and 2000 IPS PIM are installed at remote site, and connected to a 2000 IPS or IPSDM at main site over IP network, the Main Site system controls and maintains the remote DM and PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

Reserve Power

This feature provides backup power from a 24V battery source in the event of a commercial power failure.

Resident System Program

This feature provides the installers a simple procedure to have the system generate system data according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card slot location) and assigns system data (such as extension numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

Return Message Schedule Display

This feature permits any station user to register his Return Schedule from his phone when he leaves his desk or the premises, and have the Return Schedule displayed on a calling Multiline Terminal with a Liquid Crystal Display (LCD) during his absence.

Room Cutoff
This feature allows the following types of terminals to temporarily restrict guest room telephones from making unauthorized calls when guests are away from their rooms. This feature allows the same restriction when the rooms are in Check Out status:

- Attendant Console
- Hotel/Motel (H/M) Front Desk Instrument
- Property Management System (PMS) terminal
- Guest room telephones using a special access code

There are two types of Room Cutoff conditions depending on the type of calls restricted.

- External Call Restriction: All outgoing calls from guest room stations are restricted in the Room Cutoff status. (Only internal calls are available.)
- Toll Call Restriction: All toll calls from guest room stations are restricted during Room Cutoff status. (Internal and local calls are available.)

Room Status
This feature provides the Hotel/Motel (H/M) Front Desk Instrument with a visual display of the guest's room status. A supplementary print out (individual and summary) can be provided.

Route Advance
This feature automatically routes outgoing calls over alternate facilities when the first choice trunk group is busy. Users select the first choice route by dialing the corresponding access code, and the equipment then advances through alternate trunk groups only if the first choice is busy.

Save and Repeat
This feature allows a Multiline Terminal to save a specific dialed number and then redial that number at a later time.

Security Alarm
This feature provides an indication on the Attendant Console when a contact closure occurs.

Set Relocation
This feature enables two stations to be moved from one location to another without reprogramming station data at MAT.

Single Digit Dialing
This feature provides the station user the ability to dial single digit codes to access certain features while still allowing the same digit dialed to be used as the first digit of guest room station numbers.

SNMP
Simple Network Management Protocol (SNMP) is a standard protocol for TCP/IP network management, which enables a network management application software to query a management agent (network device such as router, PC host, and hub) using a supported MIB (Management Information Base). The MIB is a database of network performance information that is stored on the network devices. The NEAX 2000 IPS can support the SNMP standard MIB (MIB-II, defined in IETF RFC 1213) and private MIB and TRAP. This feature also enables the network management system (SNMP manager) to manage the 2000 IPS via Network Address Translation (NAT).

Software Line Appearance (Virtual Extensions)

This feature permits assignment of circuits which do not physically exist, to be used as secondary extensions on Multiline Terminals. There are 1020 software lines (minus the number of Dterms and Dterm IP) that can be assigned to line keys and used as desired.

Stack Dial

This feature enables a Multiline Terminal or an Attendant Console to remember the numbers dialed in the preceding five calls, including the last number dialed. The stack dial numbers are sequentially displayed on the LCD display, thus allowing the station user to make an outgoing call by selecting the desired dialed number from the display.

Station Hunting

Three Station Hunting arrangements are available. Station Hunting - Circular processes the call no matter which station in the hunt group is called. Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called. Station Hunting - Secretarial is initiated when a busy secretarial station in a Station Hunting - Circular group or Station Hunting - Terminal group is reached.

Station Hunting - Circular

When a busy station in a hunt group is called, this feature permits the call to be processed automatically through the hunt group in a preprogrammed order from that station's position within the hunt group.

Station Hunting - Terminal

When a pilot number is dialed and that number is busy, sequential Station Hunting will be initiated. However, if a number other than the pilot number is dialed and that number is busy, busy tone will be provided rather than initiate Station Hunting.

Station Hunting - Secretarial

This feature allows assignments to be given to members of Terminal and Circular Hunting groups to reroute calls (when their hunting group is all busy) to a back-up hunting group.

Station Message Detail Recording (SMDR)

This feature provides a call record for outgoing station-to-trunk calls and incoming trunk-to-station calls (including Data Call). This facilitates cost control by identifying trunk use and misuse by individual stations. Station Message Detail Recording (SMDR) enables call billing to customers and clients, and provides a means for checking local telephone bills.

Station Speed Dialing

This feature allows a station user to dial frequently called numbers by dialing an access code and an abbreviated code, or by depressing a feature key or line key assigned for Station Speed Dialing capability.

Step Call

This feature allows the Attendant or station user, after calling a busy station, to call an idle station by simply dialing an additional digit. This feature will operate only if the number of the idle station is identical to that of the busy station in all respects, except the last digit.

Supervisory Control of Peripheral Equipment

When various types of peripheral equipment (such as facsimiles, dictation equipment, Voice Mail, etc.) are connected to the line circuits of the NEAX2000 IPS, this feature allows the loop of the line circuit concerned to open for a programmable interval, and send a release signal to the peripheral equipment when the calling party disconnects.

System Clock Setup by Station Dialing

This feature enables a Station User to set up the System Clock, from Single Line Telephone, Multi-line Terminal, and PS.

System Speed Dialing

This feature provides all users the ability to dial frequently called numbers using an abbreviated call code.

Tenant Service

This feature provides for more than one organization (tenant) to share the same NEAX2000 IPS system. Through system programming, each organization may be restricted to its own Central Office trunks, Attendant Consoles and extension group. In addition, incoming calls are directed to the specific tenant.

Tie Lines

This feature allows any station user dial access or direct access to an E&M Tie Line.

Tie Line Tandem Switching

This feature allows trunk-to-trunk connections through the NEAX2000 IPS without the need for any Attendant assistance or control. The major use of this feature is in association with a dial tandem tie line network to allow tie line connections and incoming tie line calls automatic access to, and completion of, local Central Office calls.

Timed Queue

When a user originates an outgoing trunk call and the called party is busy or does not answer, the caller can set the Timed Queue feature. When this feature is set, the trunk seizure is repeated and the number is redialed after a predetermined time interval.

Timed Reminder

This feature allows the system to be programmed to automatically call stations at specified times. Upon answering, the station is connected to a recorded announcement or music source.

Trunk - Direct Appearances

This feature allows Multiline Terminal users the ability to access a CO line or E&M Tie Line without dialing an access code. For this feature, trunks must be assigned to the line keys on the Multiline Terminal. Incoming calls on CO lines can be answered on the appropriate trunk line appearance.

Trunk Queuing - Outgoing

This allows a station user, upon encountering a busy signal on a trunk, to dial a feature access code and enter a first-in, first-out queue. As soon as an outgoing trunk becomes available, stations in the queue will be called back on a first-in, first-out basis.

Trunk-to-Trunk Connection

This feature provides any station user with the ability to conference together two outside trunk calls and abandon the connection without dropping the Trunk-to-Trunk Connection.

Uniform Call Distribution (UCD)

The Uniform Call Distribution (UCD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the UCD group. Supervisor stations may monitor conversations of agents.

Busy In/Busy Out-UCD

This feature allows an agent in a UCD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the UCD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

Call Waiting Indication-UCD

This feature provides a visual indication when an incoming call to a UCD group is placed in queue, due to an “all agents busy” condition. An external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

Delay Announcement-UCD

This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to a UCD group. A single announcement, or two separate announcements, can be provided.

Hunt Past No Answer-UCD

This feature allows calls targeted at a UCD group to hunt past an agents station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

Immediate Overflow-UCD

This feature allows a call directed to a UCD group to immediately overflow to another UCD group, upon encountering an “all agents busy” condition.

Priority Queuing-UCD

This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters a UCD queue. When a call is a considered as priority it is placed at the beginning of the queue.

Queue Size Control-UCD

On incoming DID/Tie line calls, the system can be assigned a threshold that limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

Silent Monitor-UCD

This feature provides the UCD group supervisor with the ability to monitor a call to a UCD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

Uniform Numbering Plan (UNP) -Voice and Data

In the numbering plan for a network to be configured through the use of Tie Lines, a Uniform Numbering Plan (UNP) is employed. When UNP is employed, a station user from any PBX within the network can call a desired party by using a uniform dialing method based on the UNP.

Variable Timing Parameters

This feature gives the IPS the versatility to change timing duration using the Maintenance Administration Terminal (MAT) or the Customer Administration Terminal (CAT). All timing parameters are set initially in the Resident System Program. These timing parameters can be changed according to the customer's requirements.

Voice Guide

This feature provides a station user with an announcement that informs:

1. The result of the operation when the station user set or canceled the service feature, instead of service set tone.
2. Which service has been set to the station; such as, Call Forwarding - All Calls, Do Not Disturb or Message Waiting, when the station goes off-hook, instead of special dial tone.

Voice Mail Integration

This feature is used to interface the NEAX2000 IPS with a locally provided stand-alone type Voice Mail System (VMS). The VMS, connected to the NEAX2000 IPS single line circuit (LC), is controlled by sending/receiving DTMF signals using this LC.

The VMS’s voice mail feature can be used by accessing this VMS directly from an extension. If a station sets its call forwarding destination to the VMS, calls to this station are connected to the VMS, and the messages can be registered according to the VMS instruction. In addition, the Message Waiting lamp of the station can be turned on automatically by the VMS.

Voice Mail Private Password

Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the Voice Mail System.

Voice Mail Transfer

This feature has two functions that provide streamlined transfer access to voice mail.

1. One touch access to VMS: When an Attendant transfers an external call to a station, and if the station is busy or unanswered, the Attendant can transfer the call to a VMS by dialing “9” or by pressing a function key provided for this feature.
2. Transferring Camp-On call to VMS: When an Attendant sets Camp-On to a busy destination station for an external call, and if the destination station does not answer by predetermined time, the call can be automatically transferred to a VMS.

Voice over IP - (H.323)

Voice over IP allows the system to transmit voice conversations over a corporate Intranet using ITUT H.323 protocol.

Whisper Page

This feature allows a secretary to interrupt the boss in a private way. By pressing a feature key or dialing an Access Code, the secretary station can voice override the conversation between the boss and another party (station or trunk). When the conversation is interrupted, the boss can hear the secretary but the other party is unaware of the Voice Override.

CCIS Features

Attendant Camp-On With Tone Indication - CCIS
This feature permits the Attendant; when the desired station at another switching office is busy, to hold an in-coming call in a special waiting mode. A distinctive Camp-On tone is sent to the busy station when the Attendant sets Camp-On. When that station becomes idle, it is automatically rung and connected to the waiting trunk party.

Attendant Controlled Conference - CCIS
This feature permits an Attendant (2400 IPX) to establish a conference, through CCIS, with up to eight parties of stations and/or trunks (inside and outside parties).

Automatic Recall - CCIS
This service feature works as a time reminder. When an Attendant-handled call through CCIS remains on hold, camped-on, or ringing unanswered for a fixed interval, the Attendant is automatically alerted.

Brokerage - Hot Line - CCIS
This feature provides a ringdown connection between two stations, each using a Multiline Terminal, in different offices in the CCIS network.

Busy Lamp Field (BLF) - CCIS
This feature provides a busy status indication of the predetermined stations within the CCIS network. The visual indication is provided with a red LED associated with each DSS button on the DSS/BLF Console and Multiline Terminal. Pressing the DSS button allows a direct access to the preprogrammed station within the CCIS net-work.

Busy Verification - CCIS
This feature permits an Attendant, via the Attendant Console on the 2400 IPX or the 2000 IPS, to interrupt a busy station's call at another switching office connected through CCIS.

Call Back - CCIS
This feature provides inter-office Call Back. A station who has dialed a busy station at another office can set Call Back - CCIS by dialing a feature access code. When this feature has been set, the setting station will ring as soon as the busy station becomes available.

Call Forwarding - All Calls - CCIS
This feature permits all calls destined for a particular station to be routed to another station or to an Attendant Console, in another office in the CCIS network, regardless of the status (busy or idle) of the called station. The activation and cancellation of this feature may be accomplished by either the station user or an Attendant.

Call Forwarding - Busy Line – CCIS
This feature permits a call to a busy station to be immediately forwarded to a pre-designated station or to an Attendant Console in another office in the CCIS network.

Call Forwarding – Don't Answer - CCIS
This feature permits a call to an unanswered station to be forwarded to a pre-designated station or to an Attendant Console in another office, when the called station does not answer after a predetermined time period.

Call Forwarding - Intercept – CCIS
This feature allows calls to an inoperative number, through a CCIS trunk, to be intercepted and automatically routed to a recorded announcement informing the caller that an inoperative number was dialed and giving the Listed Directory Number for information.

Call Forwarding - Override - CCIS

This feature allows a target station user (Station A) to call a station (Station B) which has Call Forwarding – All Calls - CCIS set.

Call Processing Indication - CCIS

This feature provides visual indications of all CCIS calls being processed or waiting processing at the Attendant Console.

Call Transfer - All Calls - CCIS

This feature allows a station user to transfer incoming or outgoing Central Office, intra-office and inter-office calls to another station in the CCIS network, without Attendant assistance.

Call Transfer - Attendant – CCIS

This feature permits a station user, while connected to a CCIS network call, to transfer a call to an Attendant Console via the CCIS network.

Called Station Status Display - CCIS

This service feature provides, on the LCD display of the calling Multiline Terminal, a display of the called station status of the remote office within the CCIS network.

Calling Name Display - CCIS

This feature permits the station name of a calling or called party at another switching office, through the CCIS network, to be displayed either on a Multiline Terminal or an Attendant Console.

Calling Number Display - CCIS

This feature permits the number of a calling or called party at another switching office to be displayed either on a Multiline Terminal or an Attendant Console.

CCIS Networking via IP

This feature provides CCIS networks with Voice over IP (VoIP) or Peer-to-Peer IP capabilities to provide a converged infrastructure over corporate Wide Area Networks (WAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm over the CCIS network (IP based) on a peer-to-peer connection basis. And, the legacy terminals (TDM-based terminals) can communicate with other legacy terminals or IP Enabled Dterm terminals over the CCIS network, via IP-PAD (IP Packet Assembler/Disassembler). Voice compression of G.729a (8Kbps) and G.723.1 (5.3Kbps/6.3Kbps) is available for those connections. The CCIS Networking via Peer-to- Peer IP provides users with all TDM-based CCIS functionality, such as feature transparency, centralized management, and centralized facilities. There are two types of connections available for CCIS Networking via IP:

- **CCIS Networking via IP (Peer-to-Peer Connections Basis)** When the distant systems are 2000 IPS, the systems are connected on a peer-to-peer basis. The CCIS call control signals are transmitted between the built-in IP trunks (CCIS Handler; CCH) on the MP card, over the Local Area Networks and Wide Area Networks (LAN and WAN). For connections between IP Enabled Dterm terminals, voice signals are also transmitted over the LAN and WAN. For connections between legacy terminals, voice signals are transmitted via IP-PADs. This connection is also available when the distant systems are 2400 IPX supporting peer-to-peer connections.
- **CCIS Networking via IP (Non Peer-to-Peer Connections Basis)** When the distant systems are 2000 IVS2, the systems are connected with IP trunks [including Voice Compression Trunks (VCT)], via Time Division Switch (TDSW). Voice signals of IP Enabled Dterm terminals are transmitted via IP-PADs, while those of legacy terminals are directly connected to the IP trunks. Call control signals between the systems are also transmitted over the IP trunks. Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps / 6.3 kbps) can be provided by the IP trunks with VCT cards. This connection is also applicable when the distant systems are 2400 IPX not supporting peer-to-peer connections.

Centralized Billing - CCIS

This feature is used to collect billing information from each office within the network and to direct it to the associated center office. Billing information is then forwarded to the central billing centers via RS232C interfaces.

Centralized Day/Night Mode Change - CCIS

This feature switches the Day/Night mode of a remote office, linked to the main office (2400 IPX) via CCIS, in accordance with the Day/Night mode switching on the Attendant Console at the main office.

Centralized E911 – CCIS

This feature allows the system to transmit a calling party number to the 911 Emergency systems over CCIS tandem connection.

Consultation Hold - All Calls - CCIS

This feature permits a station user, within the CCIS network, to hold any incoming or outgoing public network or Tie Line call while originating a call to another station within the CCIS network.

Data Line Security - CCIS

This service feature allows the lines which are used for data transmission through CCIS to be protected from interruptions such as Attendant Camp - On, Busy Verification - CCIS, Executive Right of Way, and Attendant Override.

Deluxe Traveling Class Mark - CCIS

This feature provides outgoing call restrictions within the CCIS network. The following three types of restrictions are allowed:

- Deluxe Traveling Class Mark Restriction
- Route Restriction
- Numbering Restriction

Dial Access to Attendant – CCIS

This feature allows a station user to call an Attendant Console by dialing an operator call code through the CCIS network.

Digital Display - Station - CCIS

This service feature provides a display of the station number on the Attendant Console, during an Attendant-to-station connection, within the CCIS network.

Digital Display - Trunk - CCIS

This service feature provides the Attendant with a visual indication of incoming and outgoing trunk calls during an attendant-to-trunk connection within the CCIS network. Trunk Group number and trunk identification code are displayed.

Direct-In Termination - CCIS

This feature automatically routes incoming exchange calls through CCIS to a pre-assigned station in the network, without Attendant assistance.

Distinctive Ringing – CCIS

This feature provides distinctive station ringing patterns for terminated calls, through the CCIS network, so that a station user can distinguish between incoming internal and external calls.

Do Not Disturb - CCIS

This feature allows a station user to establish Do Not Disturb (DND) status on a temporary basis, during which time access to the station from CCIS calls will be denied.

Dual Hold – CCIS
This feature allows two connected Multiline Terminals to be placed on hold simultaneously over the CCIS link. This enables the held parties to answer or originate a call from a secondary extension or the idle primary extension.

Elapsed Time Display – CCIS
This feature provides an LCD which shows the duration of time that a Multiline Terminal is connected to any trunk through the CCIS network.

Flexible Numbering of Stations – CCIS
This feature allows voice and data station numbers to be assigned to any instrument in the CCIS network, based solely upon numbering plan limitations.

Hands-Free Answerback - CCIS
This feature allows a Multiline Terminal station user to respond to a voice call, through the CCIS network, without lifting the handset.

Hot Line – CCIS
This feature allows two stations, at different nodes in the CCIS network, to be mutually associated on an automatic ringdown basis through the CCIS network.

House Phone – CCIS
This feature allows selected stations to call an Attendant Console, through the CCIS network, simply by going off hook.

Incoming Call Identification - CCIS
This feature allows an Attendant to visually identify the type of service and/or trunk group which is arriving or waiting to be answered at the Attendant Console through the CCIS network.

Individual Attendant Access – CCIS
This feature permits a station user to call a specific Attendant Console, in the CCIS network, using an individual Attendant Identification Number.

LDN Night Connection – CCIS
This feature routes Listed Directory Number (LDN) calls to a pre-selected station, in the CCIS network, when the Night mode has been entered.

Link Alarm Display - CCIS
This feature provides an indication on external equipment when the CCIS link is connected/disconnected, when the system is initialized, or when the CCH is in make-busy.

Link Reconnect - CCIS
This feature provides the system connected to CCIS network with the capability to release the redundant CCIS link connection and re-connect the link within the system for efficient usage of the CCIS links.

Message Waiting Lamp Setting - Attendant – CCIS
This feature allows an Attendant, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, on a station in 2000 IPS.

Message Waiting Lamp Setting - Station – CCIS
This feature allows a station user, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, to a station in 2000 IPS with this feature.

Miscellaneous Trunk Access - CCIS
This feature provides access to all types of external and customer-provided equipment/facilities, such as Tie Line and exchange network, along with Dictation, Paging Access - CCIS and Code Calling through the CCIS network.

Miscellaneous Trunk Restriction - CCIS
This feature denies certain stations and dial-repeating tie trunks access to specific trunk groups, such as Tie Line, exchange network, Dictation or Paging Access - CCIS through the CCIS network.

Multiple Call Forwarding - All Calls - CCIS
This feature allows the last hop of a Multiple Call Forwarding - All Calls sequence to be forwarded over a CCIS network to a station in another office.

Multiple Call Forwarding - Busy Line – CCIS
This feature allows the last hop of a Multiple Call Forwarding - Busy Line sequence to be forwarded over a CCIS network to a station in another office.

Multiple Call Forwarding – Don’t Answer- CCIS
This feature allows the last hop of a Multiple Call Forwarding - Don’t Answer sequence to be forwarded over a CCIS network to a station in another office.

Multiple Console Operation – CCIS
This feature provides console operation where Attendant Consoles are installed in more than one node in the CCIS network.

Network Station Number - CCIS (FCCS)
When 2000 IPS is connected to a 2400 IPX via CCIS link, Network Station Number can be moved to other office within the network by a simple command operation from the Centralized MAT in the 2400 IPX.

Night Connection - Fixed – CCIS
This feature routes calls normally directed to the Attendant Console to a pre-selected station in another office, through the CCIS network, when the Night mode has been entered.

Night Connection - Flexible – CCIS
This feature provides an inter-office night connection service, via the CCIS network, when the calling station and the night station belong to different offices.

Outgoing Trunk Queuing - CCIS
This feature allows a CCIS network station, upon encountering an all trunk busy signal, to dial a specified access code and enter a first-in, first-out queue. As soon as a CCIS trunk becomes available, stations in the queue will be called back on a first-come, first-served basis.

Paging Access - CCIS
This feature provides dial access to paging equipment from an Attendant Console or a station, through the CCIS network.

Restriction from Outgoing Calls - CCIS
This feature automatically restricts users of pre-selected stations from placing outgoing calls and/or certain miscellaneous trunk calls through CCIS, without Attendant assistance.

Service Display - CCIS
This feature generates LCD displays on the Multiline Terminal corresponding to the various features as they are initiated.

Single-Digit Station Calling - CCIS
This feature allows the assignment of Single-Digit Station numbers.

Station-Controlled Conference - CCIS
This feature allows any station of the 2400 IPX to establish a conference among a maximum of eight parties of stations and/or trunks (inside and outside parties) of the 2000 IPS, through CCIS.

Station-to-Station Calling - CCIS
This feature permits any station user to dial another station directly, through CCIS, without Attendant assistance.

Station-to-Station Calling - Operator Assistance – CCIS
This feature allows a station user to call another station in the CCIS network, with the assistance of an Attendant Console operator.

Toll Restriction - 3/6 Digits - CCIS
This feature allows the system to be programmed to restrict outgoing calls, through CCIS, according to specific Area and/or Office Codes. This restriction is determined on the basis of a three-digit Area Code or six-digit Area and Office Code numbering plan.

Trunk Answer from Any Station - CCIS
This feature allows any station, not restricted from incoming calls, to answer incoming calls when the network is in Night mode.

Trunk-to-Trunk Restriction - CCIS
This feature allows Trunk-to-Trunk tandem restriction by caller’s information sent from each office (e.g., caller is a trunk) through the CCIS network.

Uniform Numbering Plan – CCIS
In a CCIS network, a Uniform Numbering Plan enables a station user to call any other station in the network. Two alternative numbering plans are provided. In the first plan, the station user dials any digit station number from three to eight. The location of the office is identified by the first one-, two-, or three-digit of the station number. In the second plan, the station user dials a one-, two- or three-digit office code and any digit station number from two to eight.

Voice Call - CCIS
This feature provides a voice path, through the CCIS network, between a Multiline Terminal in one office and a Multiline Terminal in another office. This path is established from the calling party to the called party’s built-in speaker. If the called party’s **MIC** lamp is on, the called party can have a conversation in hands-free.

Voice Mail Integration – CCIS
This feature allows any station user in the CCIS network to utilize the Voice Mail System (VMS) with the Message Center Interface (MCI).

Voice Mail Private Password - CCIS
Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the voice mail system via CCIS.

ISDN Features

Call-By-Call Service Selection

Services can be selected on a call-by-call basis to all channels of a single PRI interface according to applications. Services may be used on any available channel, unlike Trunk Provisioning Service, in which services are assigned to specific channels.

Called Party Recognition Service [Direct-In Termination (DIT)]

This feature provides an incoming Direct-In Termination (DIT) call via an ISDN trunk to be connected to a predetermined station. This application can be used for a station or modem.

CPN to Network - Present

This feature allows the ISDN network to be informed of the Calling Party Number (CPN) when a call originates from a terminal connected to the System.

CPN to Terminating User - Display

This feature provides a visual display of the originating station's number and sub-address information on a Multiline Terminal for incoming ISDN calls. This provides the terminal user with a quick and accurate way to identify the originating station's number (Calling Number).

DID Addressing

This feature allows incoming ISDN-PRI calls to terminate to stations, attendant console, automated attendant, etc., based on the called party number. Direct Inward Dial trunks will be terminated to programmed destinations without attendant assistance.

DID Addressing and DOD Addressing

This feature allows the system to use DID and DOD on the same B-Channels. Trunk Provisioning Service election is not required. (B-Channels can be used for DID and DOD without separating the trunk routes.)

Event-Based CCIS

This feature allows a PBX customer who does not have a tie line (or when a customer cannot use the tie line due to busy or fault of the line), to use the various CCIS feature by using the ISDN line as a CCIS virtual tie line, on the IMX to NEAX2000 IPS connection or the NEAX2000 IPS to IPS connection.

ISDN Terminal

This feature provides the system with an ISDN Terminal or Terminal Adapter (TA). ISDN Terminal to ISDN Terminal, ISDN Terminal to ISDN Trunk, ISDN Trunk to ISDN Terminal, ISDN Terminal to Single Line Telephone, ISDN Terminal to Multiline Terminal, and ISDN Terminal to PS connections are available.

MEGACOM® Access/WATS

AT&T's MEGACOM® (WATS) network, as well as WATS from other carriers, can be used.

MEGACOM® 800 Service/800 WATS Ultra WATS

AT&T's MEGACOM® 800 (Inward WATS) network, as well as 800 WATS provided by other carriers, can be used.

MULTIQUEST® /900 Service

AT&T's MultiQuest® service can be used. (It is a "900"-type service.) Also, 900 service provided by other carriers can be used.

Sub-address – Present

This feature allows a Primary Rate Interface ISDN trunk to transfer the Called Party Sub-address information to a destination ISDN station when the call is originated by the system. Dialing the called party station number and sub-address is required.

Trunk Provisioning Service Selection

Each channel of a PRI interface can be dedicated to a particular service. Services are designated to specific channels; once designated, a channel can be used only for that service.

Wireless Features

Analog PBX Interface

This feature allows a secretary to interrupt the boss in a private way. By pressing a feature key or dialing an Access Code, the secretary

Announcement - PS No Answer / Announcement - PS Out of Zone

This feature allows calls to a PS which cannot be paged in a predetermined period of time to be routed to the announcement notifying the calling party that the PS cannot answer. Announcements can be divided between Announcement-PS No Answer and Announcement-PS Out of Zone depending on the PS condition.

Call Forwarding-Not Available

When a PS is power off or out of zone, a call directed to the PS is forwarded to a VMS, and a voice mail message can be recorded to the VMS and checked from the PS. Also the VMS can page the PS automatically after the voice mail message is recorded.

CCIS Interface

This feature allows the WCS to be integrated with NEC PBXs with CCIS interface.

Calling Name Display - PS

Without answering incoming calls or hold calls terminates to the PS, the calling party’s name can be confirmed by the indications on the LCD.

Calling Number Display - PS

This feature provides a display on the LCD of a PS receiving a call, indicating the station number.

DTMF Signal Sender

This feature allows a PS user to send the DTMF signal (tone) to the called party (terminal, voice mail system, etc.) while engaged in communication.

GroupCall-Automatic Conference (6/10-Party)

This feature permits a PS user, Dterm user or Single Line Telephone user within the system to establish a conference among as many as six or ten parties. From a PS/D term /Single Line Telephone, a maximum of 9 PSs can be paged simultaneously except the conference leader. The PSs are assigned to the simultaneous paging groups as a participant by the system data beforehand.

GroupCall-2 Way Calling

This feature permits a PS/Dterm /Single Line Telephone can page a maximum of fifteen parties simultaneously except the group call leader. After one of paged parties answers, the paging becomes the 2 Way calling between the leader and the first answered party, and paging other parties stops automatically. The PSs are assigned to the simultaneous paging groups as participants by the system data beforehand.

Handover

When the signal transmission quality becomes inferior, a PS re-originates a call automatically and seizes another radio channel, making the WCS handover the call to another zone transceiver to maintain the speech quality.

Individual PS Calling

This feature allows the calling party to page the individual PS.

Last Number Redial-PS
This feature enables a PS to store the numbers dialed in the previous five calls including the last number dialed. The stack dial numbers are sequentially displayed on the LCD, allowing the station user to make an outgoing call by selecting the desired dialed number from the display.

Multi-Line Operation-PS
PS equipped with two line keys, L1 key and L2 key, and different station numbers can be assigned to each of two lines. The number assigned to L1 key of the PS is called My Line and the other number assigned to L2 key is called Sub Line. My Line and Sub Line of a PS can be assigned to appear on the Flexible Line Keys of a Dterm and the Dterm can share the PS lines.

Multi-Site Roaming
PS user can originate or receive a call in any place of a network provided by the plural PBXs which are interfaced by JT-Q931-a.

NEAX2000 IPS Wired for Wireless
This feature allows the NEAX2000 IPS to have the Wireless PBX feature. NEAX2000 IPS provides several integrated features adding on the adjunctive configuration.

Number Sharing
This feature allows the Dterm user to have a PS as Sub Station and to get service with one telephone number. In case that one user has both Dterm and PS, with this feature used, the user is not required to have separate two telephone numbers.

- When user is at his desk, a call is terminated to Dterm.
- When user leaves his desk with PS, a call is automatically terminated to PS.

In this feature, the Dterm and PS are referred as Main Station and Sub Station, respectively.

Out of Zone Indication
When a PS user moves out of the service area and the electric field strength becomes weak, this feature notifies it to the user with the warning tone and the LCD display.

Overlap Dialing
This feature allows a PS user to receive dial tone and dial the desired number to originate a call.

Preset Dialing
This feature allows a PS user to confirm the number to be dialed before originating a call.

PS Authorization
This feature is to confirm the identity of a PS to avoid an unauthorized PS from accessing the system.

PS Location Indication
This feature is when PS calls a Multiline Terminals/Attendant Console or vice versa, this feature allows indicating the location name of the ZT that is connected to the PS on the LCD of the Multiline Terminal.

PS Location Registration
This feature allows the WCS to supervise the location of each PS, upon receiving the location registration request, to allow call termination.

Q.931a Roaming over IP Trunk
PS user can originate or receive a call in any place of a network provided by multiple PBXs that communicate by JT-Q931a protocol over IP Trunks (IPT+VCT) or Peer to Peer Trunks.

Radio Channel Changeover
This feature is to supervise and changeover the channel automatically to avoid the interference and to maintain the speech quality.

Short Message Notification (OAI)
This feature enables a Short Text Message (STM), once arrived at a mail box of the STM Server (external equipment), to be automatically distributed to the addressee PS via Open Application Interface (OAI). This feature also provides “TM Full” notification on the display (LCD) of the address PS that is busy. If the PS is out of zone, the feature makes retransmission of the STM after the PS returns to the zone.

Speech Encryption
This feature protects a call from being tapped.

Speed Dial-PS
This feature allows a PS user to dial the certain frequently called numbers using two-digit abbreviated call codes.

Station Hunting - Not Available
This feature allows a call placed to a PS station, which is out of zone or power off, to be forwarded to an idle station in a hunt group. Two Station Hunting arrangements are available.

- Station Hunting - Circular processes the call regardless of which station in the hunt group is called.
- Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called.

Voice Mail Indication
When a message is mailed in the PS, an indication of the envelop icon is displayed in the LCD of the PS.

Chapter 6 System Overview

Operating Environment

Operating Condition	Temperature	Relative Humidity
Normal Operations	5°C - 30°C 41°F - 86°F	15% - 65% Non-Condensing
Short Periods**	-0°C - 40°C 32°F - 104°F	15% - 90% Non-Condensing
During Storage and Transit	-18°C - 50°C 0°F - 122°F	8% - 90% Non-Condensing
**Not to exceed 72 consecutive hours or 15 days in a year.		

Grounding Requirements

The system grounding must have a specific ground resistance and AC noise level, and is to be connected to a predetermined terminal in the PBX. Standard grounding requirements are as shown below:

- Communication grounding : Less than 10 ohm
- Protective ground for PIM : Less than 10 ohm

Note: *The AC ripple on these various grounds should be less than 0.5Vp-p.*

> **CAUTION**
> Grounding circuit continuity is vital for safe operation of telecommunication equipment. Never operate this equipment with the grounding conductor

The following specific requirements apply to ground wiring.

An equipment grounding conductor that is at least as large as the ungrounded branch-supply conductors is to be installed as part of the circuit that supplies the PBX. Bare, covered, or insulated grounding conductors are acceptable. Individually covered or insulated equipment grounding conductors shall have a continuous outer finish that is either green, or green with one or more yellow stripes. The equipment grounding connector is to be connected to ground at the service equipment.

The attachment-plug receptacles in the vicinity of the PBX are all to be of a grounding type, and the equipment grounding conductors serving these receptacles are to be connected to earth ground at the service equipment.

AC Power Requirements

DESCRIPTION	SPECIFICATIONS
AC Input Voltage	90 to 132Vac or 180 to 264Vac; 47 to 64Hz
AC Input Current	3.5A(at 100V), 2.0A(at 200V)

Installation

The following items are required for correct operation.

a.) Adequate space accommodation

b.) Adequate ventilation

c.) Commercial AC power

Main Equipment

The installation of the NEAX2000 IPS is comprised of up to 8 Port Interface Modules (PIMs). A PIM provides 13 card slot for Common Control, Line/Trunk (LT), and Application Porcessors (AP) cards. It also houses an AC/DC Power Supply, DC/DC Power Supply (for –48V), and batteries for protection from short-term (about 30 minutes) power interruption.

Cabling inside the unit, between the switching equipment and the MDF, can all be done by plug-and-jack connections, while printed circuit cards can easily be plugged into the edge connectors. On all installations, a special provision for plug-and-jack connections completely eliminates possible errors in wiring. This allows the installation to be done easily and smoothly.

Mounting Circuit Cards

(1) Before mounting the circuit cards, confirm the following items.

- Wrist Strap is connected to Frame Ground.
- Switch settings of circuit cards are already completed.
- The “SW1” switches of all PZ-PW121 cards are turned off.

(2) Mount circuit cards into their mounting positions according to the “Bay Face Layout” and “Port Assignment Table” given in the Office Data Programming Manual.

Various installation Methods

To meet the specific needs of the customer's environment, NEAX 2000 IPS provides the following installation methods:

- Floor Standing Installation
- Wall-mounting Installation
- IEC standard 19 inch Rack-mounting Installation

System Administration

In this system, the Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) is used for programming the system data. The CAT is a digital multi-function telephone (Dterm) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. The Maintenance Administration Terminal (MAT) is a personal computer that provides an interface to the PBX via the system CPU card. The MAT PC must have the MATWorX™ program properly installed to communicate with the PBX. MATWorX is required for system software registration and activation.

Password Entry

In a system with password service, a maintenance person is required to enter an authorization level number (Password Level) and appropriate password prior to engaging in programming the system data with the MAT/CAT. A maximum of eight (8) Password Levels can be set up. The number of commands that the maintenance person can access is determined by the Password Level.

Resident System Program

This resident system program generates system data automatically according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card location) and assigns the system data (such as station numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

Service Conditions

1. This service is applicable for equipment installed in PIM0 through PIM3.
2. Data for any vacant slot is not assigned.
3. Virtual stations are not assigned.
4. A line/trunk card (PN-AUCA/PN-DK00/PN-CFTA/PN-CFTB/PN-2AMP/PN-4DAT/PN-4RSTF/PN-4VCTI/PN-32IPLA) is not assigned, even if mounted.
5. An application card is not assigned, even if mounted.
6. No tenant assignment is provided. (Tenant 01 is assigned)

Maintenance

System Diagnostics

When a fault occurs in the system, an audible and visual indication will be given at the following units:

- External alarm indicating unit
- Fault messages reported at MAT*WorX* for remote reporting
- Alarm lamps in front of each package mounted in the frame

Self Diagnostic/System Messages

The NEAX 2000 IPS provides a sophisticated array of self-diagnostic routines that are continually and automatically being performed. Various system messages are printed when a fault occurs in a central processor system, switch network processors. Many other miscellaneous system messages and change of key status messages are also printed.

Customer Administration Terminal (CAT)

The Customer Administration Terminal (CAT) is a digital multi-function telephone (D^{term}) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. Programming of the system can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming and denying access to others.

Service Conditions

1. Programming from a Customer Administration Terminal can only be accomplished when the system is online.
2. All Multiline Terminals with LCD scanned during initialization will be Customer Administration Terminals.
3. The commands CM00 (Office Data All Clear) and CM01 (Office Data Partial Clear) cannot be accessed from the CAT. The CAT cannot delete itself from the system program.
4. Only two Customer Administration Terminals can be in program mode at the same time.
5. The data that can be changed from the CAT can be limited by the Password level assigned. There are eight levels of Passwords that can be assigned in system programming. The relation between Password level and access to available commands is also assigned in system programming.
6. A password can consist of a maximum of any eight digits with the following limitation: The password cannot be CCCCCCCC or FFFFFFFF.
7. Caution should be exercised when assigning Passwords to command authorization levels. If a password is forgotten, access to system programming will be limited and a system initialization with subsequent programming may be required.
8. When the Customer Administration Terminal is offline for programming, it cannot access normal terminal functions.

Maintenance Administration Terminal (MAT)

The Maintenance Administration Terminal (MAT) is a personal computer (PC) that is used for programming and maintenance of the NEAX 2000 IPS system. The MAT can provide a Maintenance Printout, Peg Count information and fault message output. Additionally, the MAT can be used to Remove and Restore to service any station in the system and can read or save system data from disks. The MAT can assign the Key Data for the Attendant Console. The MAT requires an IBM or compatible PC running Microsoft Windows 98, NT 4.0, 2000 or XP and MAT*WorX*.

MATWorX is a Graphical User Interface (GUI) program that provides an efficient method for manipulating the PBX database. This program contains extensive help files, Usage Wizards and Tool Tips, with hyperlinks imbedded in the text. The hyperlinks provide quick access to the appropriate Add-In modules. Add-In modules provide a user-friendly, intuitive method for customizing the PBX database.

MATWorX add-ins makes it easy for you to add or remove PBX features at any time. Add-ins are modular components that let you program specific features such as Caller ID, Station Assignments, Day/Night Modes, and Line Key Assignments.

Because add-ins are modular, you can add, remove, and upgrade them individually from within MATWorX. Add-ins let you modify your PBX's features without having to upgrade the MATWorX application itself. MATWorX also gives you a convenient way to launch other commonly used applications, such as Microsoft Word or Excel, directly from its Toolbar.

Connecting to a PBX

There are three ways to connect your PC to an NEC PBX:

- Use a modem to establish a dial-up connection.
- Use a serial cable to establish a direct connection.
- Use TCP/IP over your Local Area Network (LAN), Requires DeviceServerWork (DSW)

The method you use depends on how you installed and configured the device to which you want to connect. A serial cable direct connection offers better performance than a modem connection, but requires that the PC and device be within 50 feet of each other. A TCP/IP connection offers excellent performance and flexibility but requires a network connection to both your PC and the device.

PBX Configuration Wizard

The PBX Configuration Wizard is a custom tool in MAT*WorX* that enables you to establish the proper communication settings between your computer and the NEAX2000 IPS. The Wizard asks you simple questions and then uses the information to automatically configure the connection for the PC and the PBX.

Service Conditions

1. Connection through modems is available, providing remote maintenance capabilities.
2. MAT*WorX* can be connected to the system either directly or remotely. Direct connection is through the RS connector on the MP card. Remote connection is available via either an internal modem on the MP card or an external modem for high speed. Remote connection via the internal modem is through the COT card. Connection between the modem and the COT is accomplished through internal switching of the TDSW. Remote connection via an external modem is through the MP card.
3. The following functions can be performed from MAT*WorX*:
    - System, station, and trunk data entry, change, and copy.
    - Loading, saving, and verification of system data to and from a disk.
    - ROM check readout of generic program.
    - Display of fault/fault cleared messages.
    - On-site or remote access to the system.
    - Printout of system data (when printer is connected to PC).
    - Display and setting of system clock/calendar.
    - Numbering Plan
    - Least Cost Routing (LCR)
    - System initialize
    - UCD/Station Hunting/Call Pickup – Group
4. The PC used with MAT*WorX* must have an RS-232C port, and cannot be located more than 50 feet (15m) from the system when connected on premises.
5. When stations or trunks are expanded, moved, or changed, office data for a Multiline Terminal key/station/ trunk can be copied and multiple assignments of related office data is possible.

MACH Script Editor

This is a powerful timesaving tool that enables you to create a list of NEAX2000 IPS commands that perform tasks in the PBX. This list is referred to as a script, which can be saved and run at anytime. You can also use the MACH Script Editor to perform many other operations.

Remote Maintenance

This feature allows station and trunk changes or reassignments to be performed without a site visit by service personnel, and can be used to retrieve fault codes prior to visiting a site. One Remote Maintenance center can service an unlimited amount of systems, thus reducing the amount of personnel to maintain each site, travel costs and customer billing for each site.

Service Conditions

1. The following additional equipment is required for this feature:
   - A modem at the maintenance center and one at each remote site. (When the internal modem of the Main Processor (MP) is used, no modem at each remote site is required)
   - A cable for connection between the MP and the on-site modem. (When the internal modem of the MP is used, the above cable is not required)

2. The internal modem of the MP is compatible with the following specifications:

ITU-T V.22 1200 bps	ITU-T V.32 4800/9600 bps
ITU-T V.22 bis 2400 bps	ITU-T V.34 19.2 k/33.6 kbps
Bell 212A 1200 bps	

3. Any one of the following connections are also required for access to the modem:

A dedicated line	Attendant controlled transfer
Direct Inward System Access (DISA)	Direct Inward Termination

4. The following operations can be executed from the Remote Maintenance location:
   - Retrieval of fault data
   - Retrieval of Peg Count information
   - Deletion or addition of system data (line, trunk, etc.) using a preprogrammed security password
   - Data assignment by device number (stations, trunks, and Attendant Console)
   - Copying of station data from one station to other stations (when adding sequential stations in groups)
   - Release / Reconnection of backup batteries
   - Display of station line status

NEAX 2000 IPS Documentation List

NEC offers a full complement of documents for the NEAX 2000 IPS product line. Technical documentation is available on Compact Disk (CD ROM) or on the WEB through NTAC On-Line. This section lists all documents included on the Compact Disk (CD ROM).

NEAX 2000 IPS Technical Manuals	
Call CenterWorX Business System Manual	NEAXMail AD-8 Console Maintenance Manual
Call CenterWorX MIS Admin Manual	NEAXMail AD-8 EasyMade Application Maunal
Call CenterWorX MIS Installation Manual	NEAXMail AD-8 Reference Manual
Call CenterWorX MIS Reports Manual	NEAXMail AD-8 System Manual
CCIS System Manual	NEAXMail IM 16 System Manual
Command Manual	OAI System Manual
Configuration Guide	Office Data Programming
Data Interface System Manual	Q-SIG System Maunal
DM Hardware Installation Guide	Q-SIG System Manual (PRT)
DRS Installation and Configuration Guide	Remote PIM System Manual (Digital Remote)
Feature Programming Manual	Retrofit System Guide
General Description	SMDR/MCI/PMS Interface Specification
INASET Installation Guide	SNMP Implementation
In-Skin Router Installation Guide	System Manual
Installation Procedure Manual	Upgrade Guide
ISDN System Manual	WCS System Manual (PCS)
Maintenance Manual	Wireless System Manual
MatWorX Installation Guide	

NEAX 2000 IPS Features and Specification	
Business Hotel Features and Specifications	ISDN/Q-SIG Features and Specifications
CCIS Features and Specifications	WCS Features and Specifications
CCWX ACD Features and Specifications	

NEAX 2000 IPS User Manuals	
Dterm Assistant User Guide	MatWorX User Guide
Dterm Series E Agent Console User Guide	MatWorX Studio User Guide
Dterm Series E Supervisor Console User Guide	NEAXMail AD-8 User Guide
Dterm Series E User Guide	Power Patch Panel User Guide
Dterm Series I User Guide	Request for Proposal
Dterm Series IP User Guide	SN716 Desk Console User Guide
INASET for IPS User Guide	

Chapter 7 Specifications

Processors

The NEAX2000 IPS, IPS^{DM}, and IPS^{DMR} are distributed multiprocessor systems. Their control system consists of a Main Processor (MP), Firmware Processors (FP), and Application Processors (AP). Both the FP and APs execute their predetermined functions under the control of the MP.

DESCRIPTION	SPECIFICATIONS
Control System	Stored Program Control
Processor Type	32-Bit
Program Storage	Flash ROM
Office Data Storage	Flash ROM
Processor Architecture	Central
Program Updates	Floppy Disk
Time Division Matrix	PCM Time Division (1,024 x 1.024; Non-Blocking)

Memory

CARD NAME	PROCESSOR TYPE	MEMORY CAPACITY	
		ROM	RAM
MP (PN-CP24-A)	32-bit (133MHz)	11B (Flash ROM)	32MB (SDRAM)
FP (PN-CP15)	16-bit (25MHz)	===	768KB
AP (PN-AP00-B)	16-bit (8MHz)	512KB (Flash ROM)	512KB

Main Processor (MP)

Name Code	Remarks
PN-CP24	Main Processor Card for NEAX 2000 IPS and IPS$^{DM.}$ One card is required per system.
PN-CP27	Main Processor Card for NEAX 2000 IPS Dual MP System. One card is required per system.
PN-CP31	Main Processor Card for NEAX 2000 IPS$^{DMR.}$ One card is required for each Remote Site.
PZ-M606	Ethernet Control Card: • Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. • 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.

Major specifications and functionality of the NEAX IPS MPs are shown below:

Item	PN-CP24-B PN-CP27-A	PN-CP31-A
Central Processing	ElanSC520	
System Memory	Flash ROM (8MB), SDRAM (32MB)	
Network Switching	1,024 × 1,024 Time Division Switch	
3-Way Conference	16 sets of 3-way conference circuitry	
DTMF Signal Sender	32 circuits (digit 0 to 9, *, and # are generated)	
Music-on-Hold	10 types are available **Note 1**	
Mini Jack	1 for External Music Source for Music on Hold **Note 1**	
Audible Tone Generator (DTG)	Available	
Phase Lock Oscillator (PLO)	2 ports (Source/Receiver)	
Built-in SMDR	Available	
Built-in MCI	Available	
Built-in FP0	Available	
BS00 Function	Available	
DTMF Receiver	4 circuits	
AP01 Function	Available	
Built-in DRS	Available	
MAT Interface	---	---
Direct Connection	1 port	1 port
Remote Connection w/Built-in MODEM	1 port	Not Available
External Alarm Indication	MJ and MN	MJ only
DAT	2circuits (120 seconds per circuit)	Not Available
DK00	2 circuits (relay drive x1, external key scan x1)	Not Available
Application Key Program	In EPROM	In Flash ROM
	Built-in DRS (Device Registration Server)	

Firmware Processor (FP)

Firmware Processors (FP) are required when more than two PIMs/Modular Chassis (MC) are used. The FP provides supervision and status analysis of line/trunk ports, which reside in the MC or PIM. The FP provides the bus interface for I/O Bus, PCM Bus, and Alarm Bus in a multiple-PIM configuration. The major specifications of the FP are shown below:

- Central Processor Unit: 16-bit (25 MHz)
- Memory: Program Area (384 kb), Work Area (384 kb)
- BS01 Function

Name Code	Remarks
PN-CP15	Firmware Processor Card for use with the NEAX 2000 IPS.
PN-CP19	Firmware Processor Card for use with the NEAX 2000 IPS^{DM}.

Power

AC Power Requirements

DESCRIPTION	SPECIFICATIONS
AC Input Voltage	90 to 132Vac or 180 to 264Vac; 47 to 64Hz
AC Input Current	3.5A(at 100V), 2.0A(at 200V)

AC Power Consumption / Thermal Output (Maximum)

DESCRIPTION	AC Power Consumption (KVA)		Thermal Output (BTU)	
	100V	200V	100V	200V
1-PIM	0.35	0.40	1,195	1,365
2-PIM	0.70	0.80	2,389	2,730
3-PIM	1.05	1.20	3,584	4,096
4-PIM	1.40	1.60	4,778	5,461
5-PIM	1.75	2.00	5,973	6,826
6-PIM	2.10	2.40	7,167	8,191
7-PIM	2.45	2.80	8,362	9,556
8-PIM	2.80	3.20	9,556	10,922

Battery Requirements

DESCRIPTION	SPECIFICATIONS
Max. Battery Capacity	260AH per 4 PIM (65AH (12V) x 8)
DC Input Voltage for Battery	-24V
Built-in Battery Requirements	3.4AH (12V) x 2 (approx. 30min. backup)
Physical Size of Built-in Battery (one 12V battery)	133(W) x 60(H) x 67(D) mm

Operating Environment

DESCRIPTION	SPECIFICATIONS
Ambient Temperature	0 to 40□
Relative Humidity	Max. 90% (non-condensing)

Electrical Characteristics (Central Office Trunk)

DESCRIPTION	SPECIFICATIONS
Insulation Resistance	15 mega-ohms or more at 100Vdc
DC Resistance	On-hook conditions: 30 mega-ohms Off-hook conditions: 1,700 ohms
Impedance	On-hook conditions: 20 kilo-ohms (300 to 3,400Hz) 8 kilo-ohms (at 24Hz) Looped conditions: 600 ohms
Leak Current	0 mA at on-hook conditions

Transmission Characteristics (For TDM Circuits)

DESCRIPTION	SPECIFICATIONS
PCM Coding System	A-law/U-law
Insertion Loss	0.15 dB at 1KHz
Return Loss	20 dB or more (300 to 3,400Hz) against 600 ohms
Longitudinal Balance	59 dB or more (300 to 3,400Hz)
Attenuation/Frequency Distortion	-0.2 dB to +0.7 dB (300 to 3,400Hz)
Group Delay Distortion	0 to 0.3msec. (500 to 2,800 Hz)
Total Distortion	25 dB (Input signal:-45 dBm0)/40 dB (input signal:0 dBm0)
Idle Channel Noise	-67 dBmop or less (psophometric noise) -50 dBm0 or less (single frequency noise)
Impulsive Noise	0 counts at -35 dBm
Cross Talk Attenuation	90 dB or more
Inter-modulation Products	-40 dB or more
Spurious In-Band Signals	-49 dBm0 or less
Signal Attenuation	Attenuation rate: 12 dB per octave or more at 3.4 kHz above Attenuation level: -40 dBm or less at 3.4 kHz and above -70 dBm or less at 50 kHz and above

System Capacity

IPS System Capacity (Single MP System)

Item		Capacity Per PIM Note 1							
		1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
LT Card	(No. of Ports)	64	128	192	256	320	384	448	512
	(No. of Cards)	12	24	36	48	60	72	84	96
AP Card	(No. of Ports)	Max. 256 ports per system							
	(No. of Cards)	12	24						
Total Number of Lines (Analog Single Line Tel. + D^{term})		64	128	192	256	320	384	448	512
IP PAD	(No. of Channel)	64		128		192		256	
Analog Single Line Telephone (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	48	96	144	192	240	288	336	384
D^{term} (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	24	48	72	96	120	144	168	192
D^{term} IP/D^{term} IP INASET (PTP Connection)		956	892	828	764	700	636	572	508
D^{term} PS		512							
Cell Station (CS) / Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
ISDN Station		16	32	48	64	80	96	112	128
Central Office Trunk (Lines)	Loop Start	64	128	192	256	256	256	256	256
	DID w/4DIT	48	96	144	192	240	256	256	256
Tie Line Trunk (Lines)	2W E&M	24	48	72	96	120	144	168	192
	4W E&M	24	48	72	96	120	144	168	192
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M	DTI: 10, CCIS: 8							
	2MI	8							
ISDN	1.5M/2M (PRT)	8							
	2BRT (card)	12	24						
	4BRT (card)	6	12	18	24	24	24	24	24
IP Trunk		1	2	3	4	5	6	7	8
PFT Connections		8	16	24	32	40	48	56	64
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference group per system							
Built-in Router		Max. 8 cards per PIM							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					
Attendant Consoles		8							

IPS System Capacity (Single MP System cont'd)

Item	Capacity Per PIM Note 1							
	1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Attendant Terminal (D^{term} ATT Position)	Max. 8 sets per system							
SMDR Interface	Max. 1 interface port per system							
PMS Interface	Max. 1 interface port per system							
ACD / MIS or OAI Interface	Max. 1 interface port per system							
Remote PIM over IP (Number of PIM at Remote Site)	Up to 15 (depending on network)							
DID Dial Conversion	1000							
Call Forwarding-Outside Set	496							
Authorization. Code / Forced Account Code / Remote Access to System(DISA)Code	3000							
Message Reminder Set	1024							
Name Display / Guest Name Display	512							
Speed Calling-Station (Station Speed Dial) Set	10000							
MP built-in SMDR Call Record	1280							

Note 1: *System Capacity is for Main site only. For Total System Capacity see IP Remote Network System Capacity.*

IPS System Capacity (Dual MP System)

Item		Capacity Per PIM Note 1							
		1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
LT Card	(No. of Ports)	64	128	192	256	320	384	448	512
	(No. of Cards)	11	23	35	47	59	71	83	95
AP Card	(No. of Ports)	Max. 256 ports per system							
	(No. of Cards)	11	23	24					
Total Number of Lines (Analog Single Line Tel. + D^{term})		64	128	192	256	320	384	448	512
IP PAD	(No. of Channel)	64		128		192		256	
Analog Single Line Telephone (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	44	92	140	188	236	284	332	380
D^{term} (Lines)	Standard	64	128	192	256	320	384	448	512
	Long	22	46	70	94	118	142	166	190
D^{term} IP/D^{term} IP INASET (PTP Connection)		956	892	828	764	700	636	572	508
D^{term} PS		512							
Cell Station (CS) / Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
ISDN Station		16	32	48	64	80	96	112	128
Central Office Trunk (Lines)	Loop Start	64	128	192	256	256	256	256	256
	DID w/4DIT	44	92	140	188	236	256	256	256
Tie Line Trunk (Lines)	2W E&M	22	46	70	94	118	142	166	190
	4W E&M	22	46	70	94	118	142	166	190
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M-AMI	DTI: 10, CCIS: 8							
	2M-AMI	8							
ISDN	1.5M/2M (PRT)	8							
	2BRT (card)	11	23	24					
	4BRT (card)	6	12	18	24	24	24	24	24
IP Trunk		1	2	3	4	5	6	7	8
PFT Connections		8	16	24	32	40	48	56	64
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference group per system							
Built-in Router		Max. 8 cards per PIM							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					
Attendant Consoles		8							

IPS System Capacity (Dual MP System, Cont'd)

Item	Capacity Per PIM Note 1							
	1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Attendant Terminal (D^{term} ATT Position)	Max. 8 sets per system							
SMDR Interface	Max. 1 interface port per system							
PMS Interface	Max. 1 interface port per system							
ACD / MIS or OAI Interface	Max. 1 interface port per system							
Remote PIM over IP (Number of PIM at Remote Site)	Up to 15 (depending on network)							
DID Dial Conversion	1000							
Call Forwarding-Outside Set	496							
Authorization. Code / Forced Account Code / Remote Access to System(DISA)Code	3000							
Message Reminder Set	1024							
Name Display / Guest Name Display	512							
Speed Calling-Station (Station Speed Dial) Set	10000							
MP built-in SMDR Call Record	1280							

Note 1: *Capacity is for Main site only. For Total System Capacity see IP Remote Network System Capacity.*

IPS^{DM} System Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC		
		1	2	3
LT card **Note 1**	No. of ports	40	80	120
	No. of cards	5	10	15
AP card	No. of ports	Max. 256 ports per system		
	No. of cards	5	10	15
Total number of lines (Analog Single Line Telephone + D^{term})		40	80	120
IP-PAD	No. of channel	32	64	96
Analog Single Line Telephone (Lines) **Note 2**	4LC w/RGU Card	20	40	60
	8LC	40	80	120
	Long Line	Not Available		
D^{term}(Lines)	Standard	40	80	120
	Long Line	10	20	30
D^{term}IP/D^{term}IP INASET (PTP Connection)		956	892	828
D^{term} PS		512		
Cell Station (CS) / Zone Transceiver (ZT)		12	24	36
ISDN Station		8	16	24
Central Office Trunk (Lines)	Loop Start	40	80	120
	DID w/4DIT	20	40	60
	2W/4W E&M	10	20	30
CCIS Trunk (Peer to Peer Connection)		Max. 127		
DTI/CCIS Digital Link **Note 3**	1.5M	5	DTI: 10, CCIS: 8	
	2M	5	8	
ISDN	1.5M/2M(PRT)	5	8	
	2BRT (card)	5	10	15
	4BRT (card)	5	10	15
IP Trunk		1	2	3
PFT Connections		4	8	12
3-Party Conference		Max. 16 conference groups per system		
6-/10-Party Conference	6-Party	Max. 4 conference groups per system		
	10-Party	Max. 2 conference groups per system		
32-Party Conference		5	Max. 8 conference groups per system	

IPS^{DM} System Capacity (Cont.)

Number of PHYSICAL MODULAR CHASSIS	Capacity Per MC		
	1	2	3
Built-in Router	Max. 5 cards per MODULAR CHASSIS		
DTMF Sender	Max. 32 circuits per system		
DTMF Receiver	16	32	
SN716 Desk Console	8		
Attendant Terminal (D^{term} ATT Position)	Max. 8 per system		
SMDR Interface	Max. 1 Interface port per system		
PMS Interface	Max. 1 Interface port per system		
ACD / MIS or OAI Interface **Note 4**	Max. 1 Interface port per system		
DID Dial Conversion	1000		
Call Forwarding-Outside Set	496		
Authorization Code / Forced Account Code / Remote Access to System(DISA) Code	3000		
Message Reminder Set	1024		
Name Display / Guest Name Display	512		
Speed Calling-Station (Station Speed Dial) Set	10000		
MP built-in SMDR Call Record	1280		

Note 1: *Each Modular Chassis has 24 Virtual LT Ports that can only be used to expand the PAD channels from 8 to 32 using the 8IPLA w/24IPLA.*

Note 2: *When 8LC card is used, the 4LC w/RGU is required which does* ***not*** *provide Message Waiting indicator.*

Note 3: *The total number of trunk line and DTI channel shall be 256 or less.*
(Each trunk line and DTI channel are required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)

Note 4: *ACD / MIS and OAI are mutually exclusive.*

IP Remote Network Capacity

Total System Capacity (Main plus Remote)

Item		Capacity
LT Ports		1020
AP Ports		256
Analog Single Line Tel. + D^{term}		980
IP PAD (No. of Channel)		256
D^{term} IP/D^{term} IP INASET (PTP Connection)		952
D^{term} PS		512
Cell Station (CS) / Zone Transceiver (ZT)		128
ISDN Station		128
Central Office Trunk (Lines)		256
Tie Line Trunk (Lines)	2W/4W E&M	192
CCIS Trunk (Peer to Peer Connection)		127
DTI/CCIS Digital Link	1.5M/2M	DTI: 10/CCIS: 8 Links
ISDN	1.5M/2M (PRT)	8
	2BRT (card)	24
	4BRT (card)	24
IP Trunk		8
PFT Connections		64
3-Party Conference		Max. 16 conference groups
6-/10-Party Conference	6-Party	Max. 4 conference groups
	10-Party	Max. 2 conference groups
32-Party Conference		Max. 8 conference groups
Built-in Router		1 per Site
DTMF Sender/Receiver		Max. 32 circuits
Attendant Consoles		8
Attendant Terminal (D^{term} ATT Position)		Max. 8 sets
SMDR Interface		Max. 1 interface port
PMS Interface		Max. 1 interface port
ACD / MIS or OAI Interface		Max. 1 interface port
Remote PIM over IP		Up to 15 (depending on network)
DID Dial Conversion		1000
Call Forwarding-Outside Set		496
Authorization Code / Forced Account Code / Remote Access to System(DISA)Code		3000
Message Reminder Set		1024
Name Display / Guest Name Display		512
Speed Calling-Station (Station Speed Dial) Set		10000
MP built-in SMDR Call Record		1280

IPS^{DMR} Capacity

Number of PHYSICAL MODULAR CHASSIS		Capacity Per MC 1	Capacity Per MC 2
LT card **Note 1**	No. of ports	40	80
	No. of cards	5	10
AP card	No. of ports	Max. 256 ports per network	
	No. of cards	5	10
IP-PAD	No. of channel	32	64
Analog Single Line Telephone (Lines) **Note 2**	4LC w/RGU Card	20	40
	8LC	40	80
D^{term}(Lines)	Standard	40	80
	Long Line	10	20
D^{term}IP/D^{term}IP INASET (Peer to Peer Connection) **Note 3**		128	
Central Office Trunk (Lines)	Loop Start	40	80
	DID w/4DIT	20	40
	2W/4W E&M	10	20
DTI	1.5M	5	10
ISDN	1.5M(PRT)	5	8
	4BRT (card)	5	10
PFT Connections		4	8

Note 1: *Each Modular Chassis has 24 Virtual LT Ports that can only be used to expand the PAD channels from 8 to 32 using the 8IPLA w/24IPLA.*

Note 2: *When 8LC card is used, the 4LC w/RGU is required which does* **not** *provide Message Waiting indicator.*

Note 3: *Remote PIMs Support up to 2 Virtual PIMs for assignment of D^{term}IP/D^{term}IP INASET only.*

IPS PIM MD (As Remote PIM) Capacity

Number of PHYSICAL PIMS		Capacity Per PIM	
		1	2
LT card	No. of ports	64	128
	No. of cards	8	16
AP card	No. of ports	Max. 256 ports per network	
	No. of cards	12	24
IP-PAD	No. of channel	32	64
Analog Single Line Telephone (Lines)	8LC	64	128
D^{term}(Lines)	Standard	64	128
	Long Line	24	48
D^{term}IP/D^{term}IP INASET (Peer to Peer Connection) **Note 1**		128	
Central Office Trunk (Lines)	Loop Start	64	128
	DID w/4DIT	48	96
	2W/4W E&M	24	48
DTI	1.5M	10	
ISDN	1.5M(PRT)	8	
	4BRT (card)	6	12
PFT Connections		8	16

Note 1: *Remote PIMs Support up to 2 Virtual PIMs for assignment of D^{term}IP/D^{term}IP INASET only.*

System Capacity (Cont'd)

- **CCIS (p-p/p-mp) and Peer-to-Peer**

Payload Size	G.729a	G.711	G.723.1
20 ms	8 Channel	8 Channel	------
30 ms	16 Channel	16 Channel	16 Channel
40 ms	16 Channel	16 Channel	------

- **VoIP (H.323)**

Payload Size	G.729a	G.711	G.723.1
20 ms	6 Channel	5 Channel	---
30 ms	8 Channel	7 Channel	8 Channel
40 ms	12 Channel	10 Channel	---

- **Payload size for Virtual IPT**

Payload Size	Max. Voice Channels Per IPT		
	G.729a	G.711	G.723.1
10 ms.	4	4	–
20 ms.	8	8	–
30 ms.	16	16	16
40 ms.	16	16	–

IP Specifications

<table>
<tr><th colspan="2">Item</th><th>Specifications</th><th>Remarks</th></tr>
<tr><td colspan="2">Voice Encoding</td><td>G.729a
G.723.1 (5.3 k/6.3 k)
G.711</td><td>8 kbps CS-ACELP
MP-MLQ/ACELP
64 kbps PCM</td></tr>
<tr><td colspan="2">IP-PAD</td><td>32 channels per card
Automatically seized per call</td><td></td></tr>
<tr><td colspan="2">FAX Communication Feature</td><td>FAX Relay Method (T.30) IP</td><td>PAD card is required.
G3 FAX (up to 14.4 kbps)
Super G3 Reciprocal: Not allowed</td></tr>
<tr><td colspan="2">DTMF Signal</td><td>H.245</td><td>H.323 IPT/IP-PAD/DtermIP</td></tr>
<tr><td colspan="2" rowspan="4">Inter-office/Intra-office Signaling</td><td>H.245</td><td>D^{term}IP to D^{term}IP connection
D^{term}IP to IP-PAD connection</td></tr>
<tr><td>PROTIMS over IP</td><td>D^{term}IP to NEAX 2000 IPS connection</td></tr>
<tr><td>CCIS over IP</td><td>Point to Multipoint connection</td></tr>
<tr><td>H.323</td><td>H.323 IPT card and IP-PAD card are required</td></tr>
<tr><td colspan="2">Jitter Control</td><td>Dynamic Jitter Buffer</td><td></td></tr>
<tr><td colspan="2">QoS
(Quality of Service)</td><td>• TOS, IP Precedence
• DiffServ</td><td></td></tr>
<tr><td colspan="2">LAN Interface</td><td>10BASE-T/100BASE-TX</td><td>Auto Negotiation is available.
100BASE-TX is recommended.</td></tr>
<tr><td colspan="2">Echo Canceller
(IP-PAD)</td><td>G.168</td><td></td></tr>
<tr><td rowspan="2">Payload Size</td><td>DtermIP/CCIS
Virtual IPT</td><td>10 ms-40 ms
(G.723.1: 30 ms unit)</td><td>Max. voice channels per card
10 ms: 12 ch
20 ms: 20 ch
30 ms: 30 ch
40 ms: 32 ch</td></tr>
<tr><td>H.323 IPT</td><td>20 ms.-40 ms. (10 ms. increments)
(G.723.1: 30 ms. fixed)</td><td>Maximum voice channels per card
G.729a G.711 G.723.1
20 ms.: 6ch 5ch -
30 ms.: 8ch 7ch 8ch
40 ms.: 12ch 10ch -</td></tr>
<tr><td colspan="2" rowspan="2">PAD Control</td><td>0 dB to +16 dB (+2 dB unit)
0 dB to –16 dB (–2 dB unit)</td><td>Setting is available per Location No.</td></tr>
<tr><td>0 dB to –16 dB</td><td>For connection via the IPT card</td></tr>
</table>

NEAX IPS^{DM}/IPS^{DMR} System Specifications

Item	Specifications
System Capacity	LT ports: Max. 40 ports / MODULAR CHASSIS, - (Max. 64 ports including 24 virtual LT ports/MODULAR CHASSIS) - Max. 120 ports / system (IPS^{DM}), Max. 80 ports / system (IPS^{DMR}) - AP ports: Max. 256 ports / system - IP ports: Max. 956 ports (IPS^{DM}), Max. 128 ports / system (IPS^{DMR}) - Card slots: 6 slots / MODULAR CHASSIS (including 1 slot for MP/FP card)
Circuit Card Mounted in MODULAR CHASSIS	All LT/AP cards of the NEAX 2000 IPS can be used for the IPS^{DM}/ IPS^{DMR} with the exception of 4LLC and 2CSI cards.
Power	AC100V – 240V (automatically adjusted)
Installation Method	Desk top-setting, 19” rack-mounting
Conditions	Temperature: 5°C – 40°C (when the system is operating) Humidity: 20% - 80% (when the system is operating)
Cooling	Cooling by FAN
Safety Standard	Complied with UL60950, CSA22.2 No. 950, EN60950, AS3260
EMC	Complied with VCCI Class A, FCC Part 15 Class A, EN55022 Class A, AS/NZS 3548 Class A

Line Conditions

Description		Specifications
Loop Resistance (including Telephone Set)		
Analog Standard Line Analog Long Line		Max. 600 ohms Max. 2,500 ohms (DP 10pps), Max. 1,700 ohms (DP 20pps) Max. 1,200 ohms (DTMF)
Loop Resistance (including Opposite End Resistance)		
Central Office Trunk		Max. 1,700 ohms
Tie Line Trunk (Loop Dial)		Max. 2,500 ohms
Tie Line Trunk (E&M)		Max. 900 ohms (only E-wire condition)
Cable Length Note 1		
SN716 Desk Console		
	8DLC/4DLC/2DLC Card	Max. 350 meters (Max. 300 meters for 8DLC card)
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
D^{term} Series i/E		
	8DLC/4DLC Card	Max. 200 meters (Max. 300 meters for D^{term} 8 and D^{term} 8D)
	2DLC Card	Max. 850 meters
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
DSS/BLF Console		
	4DLC/2DLC Card with AC Adapter	Max. 1,200 meters
Zone Transceiver		
	ZTII-S (for S-Interface)	Max. 1,210 meters @-48V, Max. 970 meters @-45V
	ZTII-S with AC Adapter	Max. 1,340 meters @-48V
	ZTII-U (for U-Interface)	Max. 1,210 meters (2-wire), Max. 2, 100 meters (4-wire) @-48V Max. 970 meters (2-wire), Max. 1,700 meters (4-wire) @-45V
	ZTII-U with AC Adapter	Max. 3,950 meters

Note 1: *Cable length is based on cable with 0.5mm diameter and without lightning arresters*

Zone Transceiver Line Conditions

Zone Transceiver Line Conditions

	S-Interface Line Conditions	U-Interface Line Conditions	
PBX Power Supply at –48V (Standard)	**4 Wire**	**2 Wire**	**4 Wire**
0.4φ	762m (2500ft)	762m (2500ft)	1310m (4300ft)
0.5φ	1219m (4000ft)	1219m (4000ft)	2103m (6900ft)
0.65φ	1676m (5500ft)	1676m (5500ft)	2895m (9500ft)
0.9φ	2438m (8000ft)	2438m (8000ft)	3958m (13000ft)
PBX Power Supply at –45V (Note)	**4 Wire**	**2 Wire**	**4 Wire**
0.4φ	609m (2000ft)	609m (2000ft)	1066m (3500ft)
0.5φ	975m (3200ft)	975m (3200ft)	1706m (5600ft)
0.65φ	1432m (4700ft)	1432m (4700ft)	2499m (8200ft)
0.9φ	2131m (7000ft)	2131m (7000ft)	3718m (12200ft)
PBX Power Supply at –43V (Note)	**4 Wire**	**2 Wire**	**4 Wire**
0.4φ	548m (1800ft)	548m (1800ft)	944m (3100ft)
0.5φ	883m (2900ft)	883m (2900ft)	1524m (5000ft)
0.65φ	1340m (4400ft)	1340m (4400ft)	2316m (7600ft)
0.9φ	1948m (6400ft)	1948m (6400ft)	3352m (11000ft)
Local Power Supply (with AC Adapter)			
0.4φ	1189m (3900ft)	3350m (11000ft)	
0.5φ	1341m (4400ft)	3958m (13000ft)	
0.65φ	1676m (5500ft)	3958m (13000ft)	
0.9φ	2438m (8000ft)	3958m (13000ft)	

Note: *Zone Transceiver is provided –48V power in normal condition. The table at –45V and at –43V show ZT Line Conditions in the event of lower voltage status.*

Traffic Capacity

Traffic Capacity

Number of PIMs	1PIM	2PIM	3PIM	4PIM	5PIM	6PIM	7PIM	8PIM
Traffic Capacity	Max. 2500 BHCA		Max. 5000 BHCA **Note**		Max. 7500 BHCA **Note**		Max. 8000 BHCA **Note**	

Note: *The traffic load of each FP shall be 2500 BHCA or less.*

DRS (Device Registration Server)

DRS (Device Registration Server)-System Based

Features of Built-in DRS		Description	Remarks
Max Registration Terminal		512/MP	
Log-in	Login without password	Not Available	Use blank as a password
	Authentication by DRS-Network Based	Not Available	
	Authentication by DRS-System Based	Available	
	Authentication by MAC Address	Available	
	Confirmation when overriding	Available	
Log-out	Dialing Log-out feature access code	Available	
	Function key	Available	
	Soft key	Available	
DHCP	Inter-working with DHCP server	Available	

Chapter 8 System Performance

Transmission Characteristics

Transmission Characteristics	
Cross Talk Attenuation	More than 70 dB at 1000 Hz
Idle Circuit Noise	Less than -65 dBm
Insertion Loss (relative to 1KHz-10 dBm)	▪ Station-to-Station - Typical 6 dB ▪ Station-to-Trunk - Typical 0.5 dB ▪ Trunk-to-Trunk - Typical 0.5 dB at 0 dB PAD control
Longitudinal Balance Trunk Side	Better than 58 dB
PCM Characteristics	▪ Line Rate - 1.544 Mbps ▪ μ-Law ▪ Meets North America TI-04 Standards
Return Loss	More than 15 dB (300 ~ 3,400 Hz)
Line Impedance	Station: 600 Ω Trunk: 600 or 900 Ω
Leakage Resistance	More than 20,000 Ω

Rotary Dial Pulse and DTMF Signaling

(1) Rotary Dial Signal

Description	Specifications	
	Receiving	Sending
Dial Speed	9 to 22 pps	10 pps +/- 0.8pps 20 pps +/- 0.8 pps
Break Ratio	55 to 77 %	67 +/- 3% or 62 +/- 3%
Inter-Digit Pause	Min. 256 msec.	300 to 1,000 msec.(10 pps)
Switch-Hook Flash Detection	384 to 2,300 msec.	Not applicable

(2) DTMF Signaling

Description	Specifications				
	Receiving	Sending			
Signal Code	High Frequency Group		1,209Hz	1,336Hz	1,477Hz
	Low Frequency Group	697Hz	1	2	3
		770Hz	4	5	6
		852Hz	7	8	9
		941Hz	*	0	#
Frequency Deviation	+/- 1.8 %	+/- 0.8 %			
Signal Duration	Min. 40 msec.	64 or 128 milli-sec.			
Inter-Digit Pause	Min. 40 msec.	32 to 240 msec.			
Signal Level	-46 to -5 dBm	-10 dBm (low group) - 8 dBm (high group)			
Unwanted Frequency Components	Not Applicable	40 dB below the power of signal frequency			

Multi-frequency Compelled (MFC) – R2 SIGNAL

(1) MFC Frequency Value

Frequencies	Forward Signals(Hz)	Backward Signals(Hz)
F0	1,380	1,140
F1	1,500	1,020
F2	1,620	900
F3	1,740	780
F4	1,860	660
F5	1,980	540

(2) MFC Combinations

Combination Number	Frequencies
1	F0 + F1
2	F0 + F2
3	F1 + F2
4	F0 + F3
5	F1 + F3
6	F2 + F3
7	F0 + F4
8	F1 + F4
9	F2 + F4
10	F3 + F4
11	F0 + F5
12	F1 + F5
13	F2 + F5
14	F3 + F5
15	F4 + F5

(3) Sender/Receiver Specifications

Description	Specifications
Sender	
Sender Transmitted Level	-8 dbm to -11.5 dBm
Frequency Variation	+/- 2 Hz
Receiver	
Sensitivity Range	-35 dBm to 0 dBm
Frequency Variation	+/- 12 dBm

Audible Tones and Ringing Signal

Audible Tones

Specifications of Audible Tone

TONE	FREQUENCY	INTERRUPTION
Dial Tone (DT)	350 Hz mixed with 440 Hz	Continuous
Special Dial Tone (SDT)	350 Hz mixed with 440 Hz	0.125 sec. ON, 0.125 sec. OFF
Busy Tone (BT)	480 Hz mixed with 620 Hz	0.5 sec. ON, 0.5 sec. OFF
Reorder Tone (ROT)	480 Hz mixed with 620 Hz	2.5 sec. ON, 0.25 sec. OFF
Howler Tone (HWT)	2,400 Hz interrupted by 16 Hz	Continuous
Service Set Tone (SST)	350 Hz mixed with 440 Hz	Continuous
Ring Back Tone (RBT)	440 Hz mixed with 480 Hz	1 sec. ON, 3 sec. OFF
Hold Tone (HDT)	480 Hz mixed with 620 Hz	0.25 sec. ON, 0.25 sec. OFF 0.25 sec. ON, 1.25 sec. OFF
Second Dial Tone	440 Hz mixed with 480 Hz	0.25 sec. ON, 0.25 sec. OFF 0.25 sec. ON, 1.25 sec. OFF
Call Waiting Ring Back Tone	440 Hz mixed with 480 Hz	1 sec. ON, 1 sec. OFF

Ringing Signal

DESCRIPTION	SPECIFICATIONS
Frequency	20 Hz (Nominal)
Voltage	75 Vrms (Nominal)
Interruption:	2 sec. ON, 4 sec. OFF (for external call) 1 sec. ON, 2 sec. OFF (for internal call)

Note: *The 2000 IPS has the capability to detect the above type of signal from Central Office and to transmit the above type of signal to PBX stations.*

Built-In Modem on MP Card

DESCRIPTION	SPECIFICATIONS
MODEM	33.6 kbps

Dimensions and Weight

NEAX 2000 IPS

Main Equipment		
	Dimensions (W x D x H: mm)	**Weight (kg)**
PIM (Fully card-mounted)	Approx. 430 x 223 x 353 (16.9” x 8.8” x 13.9”)	Approx. 11.5 (25.35 lbs)
BASE	Approx. 430 x 205.2 x 61.6 (16.9” x 8.08” x 2.43”)	Approx. 3.0 (6.61 lbs)
BASE TRAY (for UL)	Approx. 435 x 224.6 x 66.2 (17.1“ x 8.84 x 2.6”)	Approx. 1.7 (3.75 lbs)

NEAX IPS^{DM}/IPS^{DMR}

Main Equipment	Dimensions (W x D x H; mm)	Weight (kg)
NEAX IPS^{DM} NEAX IPS^{DMR}	430(W) x 365(D) x 88(H) mm	Approximately 7Kg / MODULAR CHASSIS (when all slots are occupied)

Heat Dissipation

PIM NO.	MAX. AC POWER CONSUMPTION (W/h)*	MAX BTU (BTU/h)
1	360	1226
2	720	2452
3	1080	3678
4	1440	4904
5	1800	6130
6	2160	7356
7	2520	8582
8	2880	9805